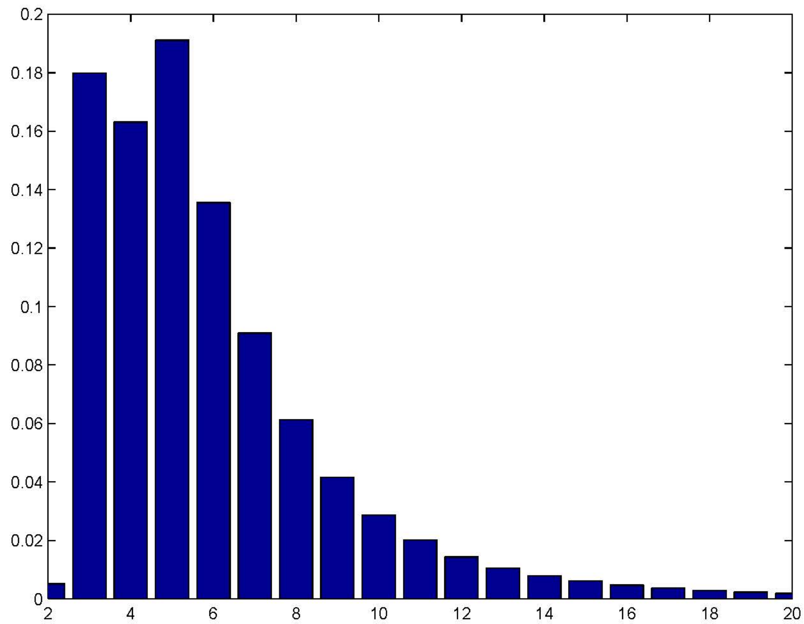


Figure 6. Histogram for needed transmissions per hybrid II ARQ packet. Both Q-O-QAM and B-O-QAM packet are included.

### *7.7.3.3 Macro cell*

#### 7.7.3.3.1 Speech

The simulated capacity is **55 kbps/MHz/cell**. Then the blocking and dropping is less than 1 % and bad quality is less than 1 %. Thus quality requirements are fulfilled with that loading. An user uses in average 1.48 slots per frame when it is active. The average fractional cell loading was 72 % in the middle cells. The used power dynamics was 10 dB.

With macro cell speech simulations the matrix was 16 time slots x 3 frequencies, which indicates reuse 1/3. In reality the number of time slots would be 64. Only 16 time slots is used in order to reduce simulation time. Thus less diversity is obtained and more blocking happens which leads to that the simulated capacity becomes lower than with the full channel matrix.

#### 7.7.3.3.2 LCD384

The simulated capacity is **113 kbps/MHz/cell.** In this simulation 98,1% of all users are satisfied, thus quality requirements are fulfilled with this loading. The used modulation is B-O-QAM and available link adaptation modes are 6/16, 8/16 and 12/16. With macro cell LCD384 simulations the matrix is 16 time slots x 9 frequencies, which indicates reuse 1/1. The average fractional cell loading is ca. 28 % in the middle cells. During the simulation, the ratio link adaptation modes (6/16, 8/16 and 12/16) were 11%, 36% and 52%, respectively.

## 7.7.4 Discussion

System simulation results for WB-TDMA are shown. The used RRM scheme is based on the interference averaging principle. In order to study the effects of fast algorithms such as ARQ and Frequency hopping, the interface between link and system level results is implemented according to [2].

The results present a lower bound case due to several reasons: downlink results are presented, all the possible enhancements by the interference averaging radio resource management algorithms are not implemented and network parameter optimization is not completed.

These simulations have been done with FDD assumption. To some extent these results can be generalized to show the trends also for the TDD mode. Detailed TDD simulations would be needed to evaluate the TDD specific features such as flexibility for resource allocation in up- and downlink and utilization of asymmetric spectrum allocations. Also the link level results for TDD may be enhanced due to algorithms that make us of reciprocity of the channel.

The future improvements to be tested include several things. Among these are interference cancellation, antenna diversity, antenna diversity, the application of ARQ to LCD data, better coding schemes (e.g. increased constraint length, optimal puncturing and concatenated codes), optimized mapping of user data into packets, fast power control and channel allocation to cells.

Future work will include the estimation of signaling overhead and degradation of the obtained capacity due to signaling. The estimated signaling load is less than 10 % of the capacity. **This load has not been subtracted from the presented capacity figures.**

The obtained simulation results show high capacity for the WB-TDMA system especially for UDD services. Main conclusions are:

- WB-TDMA with interference averaging is very promising alternative especially for non delay sensitive packet services.
- High capacities can be provided without extensive frequency and network management
- Since WB-TDMA has hardly any intra cell interference and reuse 1 can be applied, additional inter cell interference management and reduction techniques, such as adaptive antennas, joint detection and cell planning can still increase the capacity.
- For RT services the possible gains of faster quality based power control should be investigated.

## *7.8 REFERENCES*

[1] UMTS 30.03 v3.0.0 “Selection procedure for the choice of radio technologies of UMTS”, Annex 2

[2] ETSI SMG2 UMTS ad hoc #3 TDoc 73, Rennes, 1997.

[3] ETSI SMG2 Gamma Concept Group, TDoc G18/97, Sollentuna, 1997.

[4] ETSI SMG2 Gamma Concept Group, TDoc /97, Bad Salzderfurth, 1997.

# 8. High Level Requirements

This section describes how the W-TDMA concept meets the High Level Requirements for UTRA. Boxed text from ETR 04-01 has been included for reference.

Some issues are identified for further study by Gamma Group. Others may be more appropriate for consideration by SMG2 during later stages of UMTS standardisation.

## *8.1 Bearer capabilities*

### *8.1.1* Maximum user bit rate

The UTRA should support a range of maximum user bit rates that depend upon a users current environment as follows:

Rural Outdoor: at least 144 kbit/s (goal to achieve 384 kbit/s), maximum speed: 500 km/h

Suburban Outdoor: at least 384 kbps (goal to achieve 512 kbit/s), maximum speed: 120 km/h

Indoor/Low range outdoor: at least 2Mbps, maximum speed: 10 km/h

It is desirable that the definition of UTRA should allow evolution to higher bit rates.

- Rural Outdoor: 144kbps will be available throughout the operator's service area. The radio interface can tolerate the Doppler spread and rapidly changing channel characteristics associated with high speed vehicles (up to at least 1500km/h with 1/64 slot bursts). The maximum cell size depends on propagation conditions, but is comparable with GSM (assuming similar requirements for bearer capabilities and quality of service).

- Suburban Outdoor: 384kbps rate will be available with complete coverage of a suburban or urban area

- Indoor/Low range outdoor: 2Mbps will be available indoors and over localised coverage outdoors

The maximum practical bit rate which can be provided depends on factors such as the operating environment, required quality of service, traffic loading and proximity of mobile to base station.. However, the radio interface can support rates up to around 1Mbps for Rural and Suburban Outdoor, and 4Mbps over short ranges.

#### *8.1.1.1 Bearer Service Attributes*

The W-TDMA concept can provide bearers with the necessary attributes. i.e. different connection modes, symmetry, communication configuration, information transfer rate, delay variation, maximum transfer delay, maximum bit error rate, error characteristics.

### 8.1.2 Flexibility

| |
|---|
| Negotiation of bearer service attributes (bearer type, bit rate, delay, BER, up/down link symmetry, protection including none or unequal protection), |
| parallel bearer services (service mix), real-time / non-real-time communication modes, adaptation of bearer service bit rate |
| Circuit switched and packet oriented bearers |
| Supports scheduling (and pre-emption) of bearers (including control bearers) according to priority |
| Adaptivity of link to quality, traffic and network load, and radio conditions (in order to optimise the link in different environments). |
| Wide range of bit rates should be supported with sufficient granularity |
| Variable bit rate real time capabilities should be provided. |
| Bearer services appropriate for speech shall be provided. |

The W-TDMA concept provides flexibility of bearer service attributes by use of a number of different transmission bursts optimised for different bit rates in different radio environments. The Link Adaptation mechanism can be used to dynamically maintain the quality of the connection under changes in propagation and interference conditions by adjusting transmission format and number of slots allocated.

Bit rate granularity is achieved primarily by allocating different numbers of transmission slots, but this can be suplemented if necessary by adjusting channel coding rates.

Parallel bearers can be transmitted independently, or where appropriate by multiplexing together into a single channel.

Circuit switched and packet oriented services are supported efficiently by Real-Time and Non-Real-Time bearer concepts.

Variable rate data services are supported by dynamically changing the capacity allocation.

Scheduling of bearers is allowed, but could be the subject of further study by SMG2.

Bearers optimised for speech are available.

The bearer service attributes can be configured as required on initiation of a service, and changed dynamically if required.

#### *8.1.2.1 Minimum bearer capabilities*

The following table shows the potential combinations for the most important characterisation attributes (based on ETR-04-01).

| | **Real Time/Constant Delay** | | **Non Real Time/Variable Delay** | |
|---|---|---|---|---|
| **Operating environment** | **Peak Bit Rate (note 6)** | **BER / Max Transfer Delay (note 1)** | **Peak Bit Rate** | **BER / Max Transfer Delay (note 2)** |
| **Rural outdoor** **(terminal speed up to 500 km/h)** | at least 144 kbit/s granularity 13kb/s (note 3) | delay 20 - 300 ms BER $10^{-3}$ - $10^{-7}$ | at least 144 kbit/s | BER = $10^{-5}$ to $10^{-8}$ Max Transfer Delay 150 ms or more |
| **Urban/ Suburban outdoor** **(Terminal speed up to 120 km/h)** | at least 384 kbit/s granularity 74kb/s (note 4) | delay 20 - 300 ms BER $10^{-3}$ - $10^{-7}$ | at least 384 kbit/s | BER = $10^{-5}$ to $10^{-8}$ Max Transfer Delay 150 ms or more |
| **Indoor/ Low range outdoor** **(Terminal speed up to 10 km/h)** | 2 Mbit/s granularity 150kb/s (note 5) | delay 20 - 300 ms BER $10^{-3}$ - $10^{-7}$ | 2 Mbit/s | BER = $10^{-5}$ to $10^{-8}$ Max Transfer Delay 150 ms or more |

*Table 1: Minimum bearer capabilities for UMTS*

Speech bearers are supported in all operating environments.

Note 1: The minimum achievable transmission delay is less than 20ms. For a given BER operation at lower C/I is possible by extending the interleaving depth. The detailed performance trade-offs between delay and BER (via choice of modulation, coding and interleaving) require further study.

Note 2: The delivery time for NRT/variable delay bearers depends on factors such as operating environment and traffic loading. Delivery times of the order of 150ms with BER in the stated range can be provided (using Type II soft combining ARQ).

Note 3: The indicated granularity is based on BOQAM with a single 1/64 slot allocation and 1/2 rate coding

Note 4: The indicated granularity is based on BOQAM with a single 1/16 slot allocation and 1/2 rate coding

Note 5: The indicated granularity is based on QOQAM with a single 1/16 slot allocation and 1/2 rate coding

Note 6: Finer granularity can be provided by variation of channel coding rate.

### *8.1.2.2 Service traffic parameters*

Since W-TDMA provides Link Adaptation as a fundamental feature it can support the use of UMTS in various environments with a range of traffic densities range and a variety of traffic mixes in an economical way.

### *8.1.2.3 Performance*

Details of performance are given elsewhere in this evaluation document:

### 8.1.3 Configuration management

W-TDMA will allow the definition of configuration management features.

### 8.1.4 Evolution and modularity

The W-TDMA concept is service independent, and is defined so that UMTS can be implemented in phases with enhancements for increasing functionality (for example making use of different modulation and coding technology). The requirement for backwards compatibility can be met by provision of a negotiation mechanism to agree on supported capabilities between mobiles and infrastructure.

The W-TDMA concept is consistent with the requirements of an open modular architecture and implementation of software downloading of radio interface features. However these aspects require further development within SMG2.

### 8.1.5 Handover

| |
|---|
| Provide seamless ( to user ) handover between cells of one operator. |
| The UTRA should not prevent seamless HO between different operators or access networks. |
| Efficient handover between UMTS and 2nd generation systems, e.g. GSM, should be possible. |

#### *8.1.5.1 Overall handover requirements*

Efficient seamless (mobile assisted) handovers can be provided in networks with synchronised base stations and between TDD and FDD systems. Seamless handover between unsynchronised systems, is for further study by Gamma Group.

Signalling load from handovers is not expected to be significant but is dependent on scenario, and could be the subject of further study by Gamma Group.

The level of security is not be affected by handovers. Security in general is ffor further study in SMG2.

Handover to second generation systems can be supported by use of an idle frame allowing measurements of signal strengths from alternative base stations.

The choice of frame structure allows synchronisation of UTRA with GSM sharing the same cell sites which simplifies handover in this case.

#### *8.1.5.2 Handover requirements with respect to the radio operating environments*

The W-TDMA radio interface allows handovers within a network, between different environments and between networks run by different operators.

## *8.2 Operational requirements*

### 8.2.1 Compatibility with services provided by present core networks

| |
|---|
| ATM bearer services |
| GSM services |
| IP (Internet Protocol) based services |
| ISDN services |

Flexible RT and NRT bearers with a range of bit rates etc., allow current core network services to be supported.

### 8.2.2 Operating environments

W-TDMA does not restrict the operational scenario for UMTS, in, for example, international operation across various radio operating environments, across multiple operators and across different regulatory regimes. Further, a range of different MS types (e.g. speech only, high bit rate data), and a variety of services with a range of bit rates are possible.

W-TDMA can support fixed wireless access, but performance in this application is for further study in SMG2.

#### *8.2.2.1 Support of multiple radio operating environments*

W-TDMA can support the requirements of all the specified radio operating environments.

#### *8.2.2.2 Support of multiple equipment vendors*

Minimum specification levels to ensure inter-operability are for further study.

### 8.2.3 Radio Access network planning

| |
|---|
| If radio resource planning is required automatic planning shall be supported |

The Interference Averaging feature means that network planning is not as sensitive as in GSM. Detailed planning procedures are for further study in SMG2.

DCA can be used to re-configure the use of assigned frequency blocks in response to changing traffic.

UMTS terminals using W-TDMA will almost certainly incorporate frequency agility capability to support frequency hopping. This could facilitate the use over non-overlapping allocations across regions or countries (unless other hardware restrictions apply).

### 8.2.4 Public, Private and residential operators

| |
|---|
| It shall be possible to guarantee pre-determined levels of quality-of-service to public UMTS network operators in the presence of other authorised UMTS users. |

| |
|---|
| The radio access scheme should be suitable for low cost applications where range, mobility and user speed may be limited.<br>Multiple unsynchronised systems should be able to successfully coexist in the same environment.<br>It should be possible to install basestations without co-ordination.<br>Frequency planning should not be needed. |

Low cost terminals with a restricted set of functionality can be implemented. For example, limited bit rate, power output or multipath equalisation capability could be appropriate for private cordless telephone applications.

#### *8.2.4.1 Public UMTS operators*

The ability to guarantee pre-determined levels of quality for public operators is likely to require separate frequency allocations for each operator. The possibility of public operators sharing part of the spectrum is for further study in Gamma Group/SMG2.

#### *8.2.4.2 Private UMTS operators*

The Bunch concept is suitable for Private UMTS operators who may wish to operate clusters of base stations within a restricted area. A Bunch can installed without any cell planning or co-ordination with other Bunches (by using Interference Averaging), and it is co-ordination is automatically provided within a Bunch.

#### *8.2.4.3 Residential UMTS operators*

Residential systems can be deployed in the same way as private systems, except that in the limiting case there may be only one base and one mobile in the system.

## *8.3 Efficient spectrum usage*

### 8.3.1 Spectral Efficiency

| |
|---|
| High spectrum efficiency for typical mixtures of different bearer services<br>Spectrum efficiency at least as good as GSM for low bit rate speech |

The spectral efficiency is considered in detail elsewhere in this report.

Preliminary results indicate that speech and data services can be provided more efficiently than in GSM.

### 8.3.2 Variable asymmetry of total band usage

| |
|---|
| Variable division of radio resource between uplink and down link resources from a common pool (NB: This division could be in either frequency, time, or code domains) |

In an unpaired frequency allocation, TDD provides flexibility by adapting the uplink/downlink duty cycle. With paired frequency allocation, asymmetric traffic with pure FDD is likely lead to under utilisation of one or other of the band pairs. In this case TDD in the under used band would be an efficient solution. Detailed performance and consideration of other approaches is for further study in Gamma Group.

### 8.3.3 Spectrum utilisation

| |
|---|
| Allow multiple operators to use the band allocated to UMTS without co-ordination.<br>It should be possible to operate the UTRA in any suitable frequency band that becomes available such as first & second generation system's bands |

Spectrum sharing requires further study (as noted in ETR 04-01)

W-TDMA can be deployed using for some applications using a single 1.6MHz carrier (e.g. isolated cell). A small network could be deployed in as little as 5MHz (excluding guard bands). Therefore, since the concept is not critically sensitive to choice of carrier frequency this enhances the viability of deployment in any available band.

## 8.3.4 Coverage/capacity

| |
|---|
| The system should be flexible to support a variety of initial coverage/capacity configurations and facilitate coverage/capacity evolution<br>Flexible use of various cell types and relations between cells (e.g. indoor cells, hierarchical cells) within a geographical area without undue waste of radio resources.<br>Ability to support cost effective coverage in rural areas |

### *8.3.4.1 Development and implementation risk*

The W-TDMA concept is a natural extension to proven technology so unknown factors in development and implementation risks are minimised.

### *8.3.4.2 Flexibility of radio network design*

#### 8.3.4.2.1 Cell size flexibility

The W-TDMA transmission bursts are designed to cover a wide range of channel conditions. This allows operation in picocells, microcells and macrocells. Hierarchical Cell Structures are supported.

#### 8.3.4.2.2 Cell location flexibility

The Interference Averaging concept means that the system performance is not critically sensitive to base station location

### *8.3.4.3 Synchronisation*

Time synchronisation between different UMTS networks is desirable to optimise spectrum efficiency For both FDD and TDD), but is not essential.

### *8.3.4.4 Repeaters and relays*

Repeaters can be supported in principle, but the details are for further study in SMG2.

#### 8.3.4.4.1 Vehicle with mobile BS operating environment

A vehicle with mobile BS can be supported in principle, but the details are for further study in SMG2.

### *8.3.4.5 Very large cell sizes*

Very large cell sizes can be supported (for example by increasing the number of slots allocated to the bearer). Details of other techniques which could be employed, such as adaptive antennas, RF repeater stations or remote antennas are for further study.

The implications of frequency conversion of the RF carrier within a RF repeater are for further study in Gamma Group/SMG2.

### *8.3.4.6 Evolution requirements*

#### 8.3.4.6.1 Coverage evolution

The W-TDMA concept supports:

- contiguous coverage (traditional cellular approach);
- island coverage (Bunch concept);

- spot coverage (isolated cell).

Since performance is not sensitive to base station deployment, a minimum of planning is required in order to install new cells to extend system coverage. Initial calculations indicate that since maximum range (for voice) can be comparable to GSM, reusing cellsites is possible to achieve fast roll-out.

##### 8.3.4.6.2 Capacity evolution

Similarly, a minimum of planning is needed in order to install new cells to increase system capacity in areas where coverage is already provided.

W-TDMA supports techniques for capacity improvement, such as the use of adaptive antenna, but these are not essential.

## *8.4 Complexity / cost*

### 8.4.1 Mobile Terminal viability

| |
|---|
| Handportable and PCMCIA card sized UMTS terminals should be viable in terms of size, weight, operating time, range, effective radiated power and cost. |

W-TDMA is not inherently complex. Detailed complexity and other Implementation issues are for further study in Gamma Group.

### 8.4.2 Network complexity and cost

| |
|---|
| The development and equipment cost should be kept at a reasonable level, taking into account the cost of cell sites, the associated network connections , signalling load and traffic overhead (e.g. due to handovers). |

W-TDMA provides a single radio interface concept which can be adapted to all operating environments.

All the operating options within W-TDMA are based on a common approach, in order to minimise implementation complexity.

A layered approaches is has been followed in the development of the radio interface.

Detailed evaluation of various costs (e.g. migration from 2nd generation systems) is for further study in SMG2.

### 8.4.3 Mobile station types

| |
|---|
| It should be possible to provide a variety of mobile station types of varying complexity, cost and capabilities in order to satisfy the needs of different types of users. |

Mobile stations can easily be implemented with various complexity/cost/capability trade-offs. For example, low rate terminals may be not need to be capable of transmitting/receiving all possible multi-slot options. It should be possible to avoid the need for duplex filters if sufficient performance can be obtained without simultaneous transmission and reception.

## *8.5 Requirements from bodies outside SMG*

### 8.5.1 Alignment with IMT 2000

| |
|---|
| UTRA shall meet at least the technical requirements for submission as a candidate technology for IMT 2000 (FPLMTS). |

These requirements are met.

### 8.5.2 Minimum bandwidth allocation

| |
|---|
| It should be possible to deploy and operate a network in a limited bandwidth |

See section 3.2 and Annex 2.

### 8.5.3 Electromagnetic compatibility

| |
|---|
| The peak and average power and envelope variations have to be such that the degree of interference caused to other equipment is not higher than in today's systems. |

The peak power and envelope variations can be constrained so that interference is expected to be less severe than (or at least comparable) with GSM

### 8.5.4 RF Radiation effects

| |
|---|
| UMTS shall be operative at RF emission power levels which are in line with the recommendations related to electromagnetic radiation. |

Details of emission levels are for further study.

For ease of implementation, a maximum transmitter output power of around 1W peak is considered desirable for hand portable units.

### 8.5.5 Security

| |
|---|
| The UMTS radio interface should be able to accommodate at least the same level of protection as the GSM radio interface does. |

Security issues are for further study in SMG2.

### 8.5.6 Co-existence with other systems

| |
|---|
| The UMTS Terrestrial Radio Access should be capable to co-exist with other systems within the same or neighbouring band depending on systems and regulations |

W-TDMA is not inherently sensitive to co- or adjacent channel interference. It also does not produce high levels of adjacent channel interference.

Issues such as use of other systems in adjacent or the same band require further study in SMG2.

## *8.6 Multimode terminal capability*

| |
|---|
| It should be possible to implement dual mode UMTS/GSM terminals cost effectively. |

W-TDMA shares aspects of TDMA with GSM, including related frame structures. This is beneficial for implementation of dual mode terminals.

## *8.7 Services supported by the radio interface*

### 8.7.1 Location service

The detailed mechanism for support of user position location is for further study in SMG2. Since users are orthogonal in time domain in WB-TDMA, implementing position location with observed time difference can be done.

# 9. Conclusions

This draft evaluation document presents the Wideband TDMA concept and the evaluation work. The report has been prepared by the SMG2 Concept Group Gamma (WB-TDMA).

The Radio Transmission Technology (RTT) building blocks for the WB-TDMA are described in chapters for Logical Channels, Physical Channels, Layer 2 Radio Protocols and Radio Resource Management. Advanced TDMA features which can be used to enhance the performance of the WB-TDMA scheme are described. More detailed descriptions of the Layer 2 protocols and RRM schemes can be found in Tdocs SMG2 Gamma 19 and 15 respectively.

Evaluation for the services as described in Tdoc SMG2 258/97 has been done on link and system layer according to the table below.

| Priority | Environment | Service mixture | Propagation model | Cell coverage | Link level | System level |
|---|---|---|---|---|---|---|
| **1.1**<br>**1.2**<br>**1.3**<br>**1.5**<br>extra | **Outdoor to Indoor and Pedestrian 3 km/h** | **UDD 384**<br>**Speech**<br>**LCD 144 kbit/s**<br>**UDD 2048**<br>UDD 144 | **Outdoor to Indoor and Pedestrian A** | **Microcell** | **completed**<br>**completed**<br>**completed**<br>**completed**<br>completed | **completed**<br>**completed**<br>(not required)<br>(not required) |
| **2.1**<br>**2.2**<br>**2.3**<br>**2.4**<br>extra<br>extra<br>extra | **Indoor 3 km/h** | **UDD 2048**<br>**Speech**<br>**LCD 384 kbit/s**<br>**50 % speech + 50 % UDD 384**<br>UDD 2048 with walls<br>UDD 144<br>UDD 384 | **Indoor A** | **Picocell** | **completed**<br>**completed**<br>**completed**<br>(not required)<br>(not required)<br>completed<br>completed | **completed**<br>(not required)<br>(not required)<br>**preliminary**<br>completed |
| **3.1**<br>**3.2**<br>**3.3**<br>extra<br>extra | **Vehicular 120 km/h** | **UDD 144**<br>**Speech**<br>**LCD 384 kbit/s**<br>UDD 384<br>UDD 2048 | **Vehicular A** | **Macrocell** | **completed**<br>**completed**<br>**completed**<br>completed<br>completed | (not required)<br>completed<br>completed |
| extra<br>extra<br>extra | Vehicular 120 km/h | Speech<br>UDD 144<br>UDD 384 | Vehicular B | | completed<br>completed<br>completed | (not req) |
| **4**<br>extra<br>extra | **Vehicular 250 km/h** | **Speech**<br>UDD 144<br>UDD 384 | **Vehicular B** | | **completed**<br>completed<br>completed | (not req) |

Considerations how the WB-TDMA concept fulfils the SMG2 high level requirements have also been included (see chapter 8).

Specific aspects that have been considered during development of the WB-TDMA concept are:

- Support of high bit rates with relatively simple terminal
- Effective support of non-real-time traffic with fast variations in data rate and packet size
- Support of TDD mode with data rates fulfilling the UMTS requirements
- Possibility to implement simple terminals for low bit rate use
- Narrow spectrum and low interference to adjacent carriers
- Flexibility for introduction of enhancements

# 10. Annex 1 (ETR04.02)

**Technologies Description Template**

| | | |
|---|---|---|
| | A1.1 | Test environment support |
| ti | A1.1.1 | In what test environments will the SRTT operate ?<br>*All test environments described in Section 1.1 of annex 2.* |
| td | A1.1.2 | If the SRTT supports more than one test environment, what test environment does this technology description template address ?<br>*This template considers all the environments.* |
| | 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.<br>*No additional feature is required to support FWA.* |
| | A1.2 | Technical parameters<br>Note : Parameters for both forward link and reverse link should be described separately, if necessary. |
| ti | A1.2.1 | What is the minimum frequency band required to deploy the system (MHz) ?<br>*Minimum value depends on the application, the load of the network, the QoS and the user traffic model. A 2 Mbit/s is possible in an isolated cell from 1.6 MHz in TDD mode and small cellular networks are possible from 2x4.8MHz.* |
| ti | A1.2.2 | What is the duplex method : TDD or FDD<br>*Both FDD and TDD options are proposed. The TDD option allows to support asymetric traffic.* |
| ti | A1.2.2.1 | What is the minimum up/down frequency separation for FDD ?<br>*Based on DCS1800 assumptions, a sensible value seems to be the uplink or downlink bandwidth plus 10 MHz to make the duplexer feasible.* |
| ti | A1.2.2.2 | What is requirement of transmit/receive isolation ? Does the proposal require a duplexer in either the mobile or base station.<br>*In FDD mode low rate terminals can operate with non-overlapping transmission and reception time slots and in this case a duplexer is not required..* |

| | | |
|---|---|---|
| ti | A1.2.3 | Does the SRTT allow asymmetric transmission to use the available spectrum ? Characterize.<br>*Asymmetric service is possible both in the FDD and TDD modes.*<br>*It can be done easily in FDD, especially by taking profit of the unpaired frequency band, by allocating a low bit rate carrier for the uplink from the "paired" band and a high bit rate carrier from the « unpaired » band. The unused pair of the low bit rate carrier can be used for asymmetric services within the symmetric band. TDMA makes this allocation scheme reasonable to implement.*<br>*TDD asymmetry could be achieved by 'traditional' means by allocation a fixed amount of slots through the whole operator's TDD spectrum for both directions.*<br>*However, due to the uncertainty concerning traffic profiles in different environments this approach does not seem to be very attractive. One idea in the usage of the TDD band is to allocate asymmetric capacity on an individual basis i.e. the frequency band would not be divided into purely uplink and downlink channels in the time domain but rather in a dynamic manner enabling the usage of all channels in frequency and time domain for both directions. These scheme is possible both by using RNC controlled Channel Allocation which enables fast handovers in the frequency and time domain or by Interference Averaging between different cells.*<br>*The described scheme would lead to minimum frequency planning. The operator does not have to put fixed percentage of the overall system capacity to up- or downlink traffic.* |
| ti | A1.2.4 | What is the RF channel spacing (kHz) ? In addition, does the SRTT use interleaved frequency allocation ?<br>*The channel spacing is 1.6 MHz, both in FDD and TDD modes. Interleaved frequency allocation is not assumed.*<br>Note : Interleaved frequency allocation ; allocating the $2^{nd}$ 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 $2^{nd}$ adjacent channel. |
| ti | A1.2.5 | What is the bandwidth per duplex RF channel (MHz) measured at the 3 dB down points ?<br>It is given by (bandwidth per RF channel) x (1 for TDD and 2 for FDD). Please provide detail.<br>*The 3 dB bandwidth of the duplex RF channel is around 1.1 MHz in TDD mode (the exact value depending on the linearity requirements of the power amplifier which is not defined yet), while it is twice this value for the FDD mode.* |

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| | 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 ?<br>*As far as traffic is concerned, a single bandwidth of 1.6 MHz is assumed per channel.*<br>*A beacon channel (BCCH) is anticipated in addition and two options are considered to prevent the permanent broadcast of a high power signal with negative impact on the reuse pattern :*<br>*1/ discontinuous transmission of the BCCH, the bandwidth would remain 1.6 MHz*<br>*2/ continuous transmission of a narrowband signal (200 kHz). In that case, two channel bandwidths would exist in the system.* |
| ti | A1.2.6 | What is the RF channel bit rate (kbps) ? .<br>The maximum modulation rate of RF (after channel encoding, adding of in-band control signaling and any overhead signaling) possible to transmit carrier over an RF channel, i.e. independent of access technology and of modulation schemes.<br>*It is either 2.6 or 5.2 Mbit/s with the considered modulations presented below in section A1.2.11. It must be in addition mentionned that the W-TDMA concept can support any modulation scheme meeting the emission mask requirements. This will allow the use of higher order modulations to support even higher bit rate connections over short ranges (eg within a single room). It will also allow alternative methods of combatting multipath, such as OFDM. Similarly, the use of different channel coding algorithms need not be restricted.* |

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| ti | A1.2.7 | Frame Structure : Describe the frame structure to give sufficient information such as ;<br>- frame length<br>*The frame length is 4.615 ms*<br>- the number of time slots per frame<br>*There are 2 standard timeslot lengths :*<br>• *those of 72 ms, occupying a 64th of the frame length*<br>• *those of 288 ms, occupying a 16th of the frame length*<br>• *other values, within the scope of the so-called "flexible burst" introduced below are possible.*<br>*Any combination of these timeslot fitting the length of the TDMA frame is possible, including a frame of 64 short time slots and a frame of 16 long time slots.*<br>- guard time or the number of guard bits<br>*There are usually 10.5 to 11 guard symbols per burst, the length of each symbol being 0.384 ms. Three exceptions must be noticed :*<br>• *the 1/16th frame access burst with a guard period of 625 symbols,*<br>• *the 1/64th frame access burst with a guard period of 84,5 symbols*<br>- user information bit rate for each time slot<br>*It depend on the type of burst as described below :*<br>• *Data bursts, the payload is 684 symbols, resulting in roughly 296 kbit/s for the BOQAM modulation and 593 kbit/s for the QOQAM modulation.*<br>• *Speech burst 1, the payload is 122 symbols, resulting in roughly 53 kbit/s for the BOQAM modulation and 106 kbit/s for the QOQAM modulation.*<br>• *Speech burst 2, the payload is 122 symbols, resulting in roughly 53 kbit/s for the BOQAM modulation and 106 kbit/s for the QOQAM modulation.*<br>• *1/16th-frame synchronisation burst, the payload is 606 symbols, resulting in roughly 263 kbit/s for the BOQAM modulation and 525 kbit/s for the QOQAM modulation.*<br>• *1/64th-frame synchronisation burst, the payload is 78 symbols, resulting in roughly 34 kbit/s for the BOQAM modulation and 68 kbit/s for the QOQAM modulation.*<br>• *Access bursts, the payload is 56 symbols, resulting in roughly 24 kbit/s for the BOQAM modulation and 49 kbit/s for the QOQAM modulation.*<br>• *In addition, a flexible burst was introduced. The length of its different fields (tail, data symbols, training sequence and guard period) is agreed between the mobile and the base station at call setup or whenever required during a call. The actual bit rate is obviously dependant of the actual field lengths.*<br>*A multiplexed burst was defined, it is a combination of data sequences with a header describing its structure. It occupy an integer number of adjacent 1/16 timeslots, and is intended for use on the downlink.*<br>- channel bit rate (after channel coding)<br>*5.2 or 10.4 Mbit/s*<br>- channel symbol rate (after modulation)<br>*2.6 Msymb/s*<br>- associated control channel (ACCH) bit rate<br>*The rate is dynamically adjustable. It is too early to indicate its effective use.*<br>- power control bit rate.<br>*It was estimated to 0.007% of the channel capacity per mobile* |

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| ti | A1.2.8 | Does the SRTT use frequency hopping ? If so characterize and explain particularly the impact (e.g. improvements) on system performance.<br>*Frequency hopping is an option of the system to improve performances when sufficient spectrum is available to the operator.*<br>*The advantages are twofolds :*<br>*- better performances for low-velocity mobiles, as frequency hopping allows terminal to escape from a fadded frequency,*<br>*- interference diversity, it might allow a reuse factor of 1 if sufficient frequency is available.* |
| td | A1.2.8.1 | What is the hopping rate ?<br>*It is the frame rate, that is to say 216 Hz (unless the timeslot rate, 1387 Hz, is felt preferable for multislot allocation).* |
| td | A1.2.8.2 | What is the number of the hopping frequency sets ?<br>*To be Completed.* |
| ti | A1.2.8.3 | Are base stations synchronized or non-synchronized ?<br>*Base stations are normally non-synchronized. A synchronised mode might however be considered to ease monitoring of adjacent cells when discontinuous BCCH is used.* |
| ti | A1.2.9 | Does the SRTT use spreading scheme ?<br>*No* |
| td | A1.2.9.1 | What is the chip rate (Mchip/s) : Rate at input to modulator.<br>*Not applicable to TDMA.* |
| td | A1.2.9.2 | What is the processing gain : 10 log (Chip rate / Information rate).<br>*Not applicable to this SRTT.* |
| td | 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.<br>*Not applicable to this SRTT.* |
| ti | A1.2.10 | Which access technology does the proposal use : TDMA, FDMA, CDMA , hybrid, or a new technology ?<br>*FDMA / TDMA.*<br>In the case of CDMA which type of CDMA is used : Frequency Hopping (FH) or Direct Sequence (DS) or hybrid ? Characterize.<br>*Not applicable to this SRTT.* |
| ti | A1.2.11 | What is the baseband modulation technique ? If both the data modulation and spreading modulation are required, please describe detail.<br>*Two modulations are considered, depending on the bit-rate and the environment : Binary Offset QAM (B-OQAM) and Quaternary Offset QAM (Q-OQAM) with a rolloff of 0.35.*<br>What is the peak to average power ratio after baseband filtering (dB) ?<br>*To be completed.* |

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| ti | A1.2.12 | What are the channel coding (error handling) rate and form for both the forward and reverse links ? e.g.<br>- Does the SRTT adopt FEC (Forward Error Correction) or other schemes ?<br>*Circuit switched sercices can be protected by convolutional or turbo codes. Convolutional codes can be punctered to efficiently achieve adaptive coding rate.*<br>*Packet services can be protected by ARQ schemes.*<br>- Does the SRTT adopt unequal error protection ? Please provide details.<br>*This is not considered but can be supported.*<br>- Does the SRTT adopt soft decision decoding or hard decision decoding ? Please provide details.<br>*Soft decision decoding is adopted to take profit of the several dB of gain achievable by Viterbi decoding of convolutional codes.*<br>- Does the SRTT adopt iterative decoding (e.g. turbo codes) ? Please provide details.<br>*Iterative decoding is considered for Turbo codes. The number of iteration is still opened.*<br>- Other schemes. |
| ti | A1.2.13 | What is the bit interleaving scheme ? Provide detailed description for both up link and down link.<br>*Interslot interleaving is provided whenever possible. The interleaving depth can be adjusted to optimise trade-off between bearer C/I requirement and delay.* |
| ti | 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.).<br>*An equalizer is assumed to combat multipath propagation. Training sequences have been inserted in bursts for that purpose.* |
| ti | A1.2.14.1 | Describe the robustness to intersymbol interference and the specific delay spread profiles that are best or worst for the proposal.<br>*Large delay spreads can be tolerated by definition of bursts with long enough training sequences.*<br>*If there is residual ISI after equalizer, it is handled by link adaptation and ARQ. Performance degradation is graceful as delay spread increases. Also Vehicular B channel can be supported with a small degradation in performance with the current burst structure. With longer training sequences and flexible burst structure the performance could be improved.* |
| ti | A1.2.14.2 | Can rapidly changing delay spread profile be accommodated ? Please describe.<br>*Simulations show that there is no degradation for 1/64 slot up to 500 km/h. The question is still opened for other slots.* |
| ti | A1.2.15 | What is the Adjacent channel protection ratio ?<br>*To be completed.*<br>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.<br>*To be Completed.* |

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| | A1.2.16 | Power classes<br>*To be completed.* |
| ti | 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).<br>*To be completed* |
| ti | A1.2.16.1<br>.1 | What is the maximum peak power transmitted while in active or busy state ?<br>*A peak power limit of 1W is assumend for hand portables.* |
| ti | A1.2.16.1<br>.2 | What is the time average power transmitted while in active or busy state ? Provide detailed explanation used to calculate this time average power.<br>*The average power of MS depends on power control settings and number of timeslots used..* |
| ti | A1.2.16.2 | Base station transmit power per RF carrier for terrestrial component<br>*To be completed.* |
| ti | A1.2.16.2<br>.1 | What is the maximum peak transmitted power per RF carrier radiated from antenna ?<br>*To be completed.* |
| ti | A1.2.16.2<br>.2 | What is the average transmitted power per RF carrier radiated from antenna ?<br>*To be completed.* |
| ti | 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 ?<br>*64 channels can be supported per carrier of 1.6 MHz. It is however not possible at that level to precise whether G.726 performance requirements are met.* |

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| ti | 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 :<br><br>-adaptive source and channel coding as a function of RF signal quality<br><br>-variable data rate as a function of user application<br><br>-variable voice/data channel utilization as a function of traffic mix requirements ?<br><br>Characterize how the bit rate modification is performed. In addition, what are the advantages of your system proposal associated with variable bit rate capabilities ?<br>*The Radio Link Control/Medium Access Control (RLC/MAC) protocol supports two types of bearers, real time (RT) and non real time (NRT) bearers. The RT operation mode is used for the radio bearers which have strict delay constrains and quality is mainly fulfilled by power control and forward error corrections. The NRT operation mode is used for radio bearers with low delay requirements which allow backward error correction.*<br><br>*RLC/MAC layer protocol provides fast resource allocations for real time (RT) and non real time (NRT) services supporting also variable bit rates and multibearer connections. For RT services QoS is fulfilled by means of dynamic link adaptation and for NRT services QoS can be maintained by effective ARQ. Radio resources are allocated for a common pool for all bearers thus enabling immediate adaptation to any kind of traffic mix within the available resources. All bearers are controlled independently.*<br><br>*In the RT mode, the RLC entities request resources for the radio bearer due to radio condition variations and the bit rate variations. RLC resource requests are directed to MAC, which is responsible for the channel allocation signalling. Mobile initiated resource requests are transmitted on the dedicated control channel (DCCH) or random access channel (RACH). Channel allocations are transmitted on downlink DCCH or transmitted on the common control channel (CCCH). Fast associated control channel (FACCH) is a dedicated channel which uses capacity stolen from a bearer allocated to the MS. For a few occasional messages this is the preferred signalling channel. If FACCH can not be used, signalling can be transmitted on CCCH. The link adaptation is possible with appr. 9 ms intervals.*<br><br>*In the NRT mode, the RLC entities request resources for certain amount of data. For high bitrates 1/16 timeslot traffic channels are allocated for 9 ms allocation period (2 TDMA frames) at a time. Two frames gives some interleaving gain and is still very flexible. Channel allocations are announced on the NRT control channel (NCCH) and in order to avoid transmission of long identities there, a short reservation identity is allocated for the radio bearer. This identity is valid until the requested data is transmitted and during that time the mobile is obliged to listen to the NCCH. Traffic channels allocated for one reservation identity during one allocation period may vary from 0 to 14 timeslots, and the achieved bit rate may vary from 0 to 2 Mbit/s. For lower bitrates and infrequent transmissions the reservation is made from 1/16 or 1/64 timeslot traffic channels and reservation is valid for indicated time period.*<br><br>*The MAC is also responsible for ARQ signalling. CRC and reception quality based type II soft combining ARQ is expected to provide best efficiency.* |

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| td | A1.2.18.1 | What are the user information bit rates in each variable bit rate mode ?<br>*Basically the protocol is capable of supporting any bitrate restricted by the granularity of channel allocations. Currently the transmission capacity allocation granularity to RT a service is of order 200 bit/s and the smallest allocable packet size for NRT bearer is of order 250 bits (both figures are gross rates thus including channel coding).*<br>*For practical reasons it is considered currently to limit the set of possible bitrates for each RT bearer to a set of 16 alternatives, which can be freely selected to each bearer separately taking into account the above mentioned granularity. The bearer can change between these agreed bitrates dynamically during transmission. For all NRT bearers all packet sizes starting from 250 bits are available..* |
| ti | A1.2.19 * | What kind of voice coding scheme or CODEC is assumed to be used in proposed SRTT ? 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 standardised in standardisation bodies such as ITU) is indispensable for the proposed SRTT, provide detail, e.g. scheme, algorithm, coding rates, coding delays and the number of stochastic code books. |
| ti | A1.2.19.1 | Does the proposal offer multiple voice coding rate capability ? Please provide detail.<br>*The protocol for resource allocation handles multiple voice coding rates similarly as any other variable bitrate services. The realisation of multiple source coding rates is however out of the scope of the radio interface.* |
| ti | 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.<br>*TDD mode is of particular interest for asymmetric data service. The ratio of the channel allocated to uplink and downlink can be dynamically changed as a function of the traffic need.*<br>Note 1 : See [draft new] Recommendation [FPLMTS.TMLG] for the definition of<br>- “circuit transfer mode”<br>- “packet transfer mode”<br>- “connectionless service”<br>and for the aid of understanding “circuit switched” and “packet switched” data services<br>Note 2 : See ITU-T Recommendation I.362 for details about the service classes A, B, C and D |
| ti | A1.2.20.1 | For delay constrained, connection oriented. (Class A)<br>*The proposal support delay constrained connection oriented service by variable bitrate Real Time bearer, the delay constrain of which may be set to an arbitrary value (4,6 ms granularity) and which may have arbitrary bit rate variation (>250 bit/s granularity). (See A.1.2.18).* |
| ti | A1.2.20.2 | For delay constrained, connection oriented, variable bit rate (Class B)<br>*The proposal support delay constrained connection oriented service by variable bitrate Real Time bearer, the delay constrain of which may be set to an arbitrary value (4,6 ms granularity) and which may have arbitrary bit rate variation (>250 bit/s granularity). (See A.1.2.18).* |

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| ti | A1.2.20.3 | For delay unconstrained, connection oriented. (Class C)<br>*The proposed concept is capable to reliable transmission of data in any packet size with guaranteed almost arbitrary quality and unconstrained delay. Also tradeoff on the quality target and the delay requirement target can be handled. (See A.1.2.18).* |
| ti | A1.2.20.4 | For delay unconstrained, connectionless. (Class D)<br>*To be Completed.* |
| ti | A1.2.21 | Simultaneous voice/data services : Is the proposal capable of providing multiple user services simultaneously with appropriate channel capacity assignment ?<br>*Multiple timeslot allocation can provide simultaneous voice/data service.* |
| | | Note : The followings describe the different techniques that are inherent or improve to a great extent the technology described above to be presented :<br>Description for both BS and MS are required in attributes from A2..22 through A1.2.23.2. |
| ti | A1.2.22 | Power control characteristics : Is power control scheme included in the proposal ? Characterise the impact (e.g. improvements) of supported power control schemes on system performance.<br>*The scheme includes two power control options.*<br>*1) Slow power control by dedicated power control messages*<br>*-Slow power control is applicable for both uplink and downlink power control. The concept provides messages enabling both bearer specific and MS specific power control. The power control interval of this scheme can be arbitrary (>4.615 ms) or can be applied only on demand. The needed amount of power control signalling is highly dependent on the system design, i.e. how large interference variations the system is able to handle.'*<br>*2) Public Power control for uplink*<br>*-This option enables controlling the power of each physical channel separately with 4,6 ms interval. This method is only applicable for controlling the uplink powers..* |
| td | A1.2.22.1 | What is the power control step size in dB ?<br>*A step size of 0.5 dB to 4.0 dB is considered.* |
| td | A1.2.22.2 | What are the number of power control cycles per second ?<br>*Adjustable for each MS separately and can be dynamically varied during connection. See A.1.2.22.* |
| td | A1.2.22.3 | What is the power control dynamic range in dB ?<br>*A slow power control scheme is considered, with a dynamic range of 50 dB.* |
| td | A1.2.22.4 | What is the minimum transmit power level with power control ?<br>*To be Completed.* |
| td | A1.2.22.5 | What is the residual power variation after power control when SRTT 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.<br>*To be Completed.* |

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| ti | A1.2.23 | Diversity combining in mobile station and base station : Are diversity combining schemes incorporated in the design of the SRTT ?<br>*Frequency and time diversity are considered both at the base station and at the mobile, through frequency hopping, interleaving and possibly timeslot hopping.*<br>*Antenna diversity combining is possible and considered at the base station. It is possible but not considered at the mobile station.* |
| td | 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 :<br>- time diversity : repetition, RAKE-receiver, etc.,<br>- space diversity : multiple sectors, multiple satellite, etc.,<br>- frequency diversity : FH, wideband transmission, etc.,<br>- code diversity : multiple PN codes, multiple FH code, etc.,<br>- other scheme.<br>*Frequency and time hopping are considered.*<br>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.<br>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 ?<br>*One RF receiver and one antenna are considered at the mobile unit.*<br>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. |
| td | A1.2.23.2 | What is the degree of improvement expected in dB ? Please also indicate the assumed condition such as BER and FER.<br>*To be Completed.* |
| ti | A1.2.24 | Handover/Automatic Radio Link Transfer (ALT) : Do the radio transmission technologies support handover ?<br>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.<br>*Mobile assistd, seamless handover is considered.* |
| td | 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.<br>*Seamless handover is considered* |

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| td | A1.2.24.2 | For the proposed SRTT, can handover cope with rapid decrease in signal strength (e.g. street corner effect) ?<br>Give a detailed description of<br>- the way the handover detected, initiated and executed,<br>- how long each of this action lasts (minimum/maximum time in msec),<br>- the timeout periods for these actions.<br>*To be Completed.* |
| ti | 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) :<br>- in terms of frequency planning<br>- in terms of the evolution of adaptive antenna technology using mobile identity codes (e.g. sufficient number of channel sounding codes in a TDMA type of system)<br>- other relevant aspects<br>*To be Completed.* |
| ti | A1.2.26 | Sharing frequency band capabilities : To what degree is the proposal able to deal with spectrum sharing among UMTS systems as well as with all other systems :<br>- spectrum sharing between operators<br>- spectrum sharing between terrestrial and satellite UMTS systems<br>- spectrum sharing between UMTS and non-UMTS systems<br>- spectrum sharing between private and public UMTS operators<br>- other sharing schemes.<br>*To be Completed.* |
| ti | 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.)<br>*See section 5.3.* |
| ti | 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)<br>*To be Completed.*<br>Note : Cell definitions are as follows :<br>pico - cell hex radius ® < 100 m<br>micro - 100 m < ® < 1000 m<br>macro - ® > 1000 m |
| ti | A1.2.29 | Describe any battery saver / intermittent reception capability<br>*To be Completed.* |

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| td | 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.<br>*To be Completed.* |
| td | 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.<br>*The same transmission scheme is anticipated for data and signaling.* |
| td | A1.2.30.1 | Describe the different signaling transfer schemes which may be supported, e.g. in connection with a call, outside a call.<br>Does the SRTT support new techniques ? Characterise.<br>Does the SRTT support signalling enhancements for the delivery of multimedia services ? Characterise.<br>*See section 4.* |
| ti | 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, characterise these capabilities.<br>*BOD can be accommodated by multi time slot allocation.*<br>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). |
| ti | A1.2.32 | Does the SRTT support channel aggregation capability to achieve higher user bit rates ?<br>*Not considered.* |
| | A1.3 | Expected Performances |
| | A1.3.1 | for terrestrial test environment only<br>*See section 7.* |
| ti | A1.3.1.1 | What is the achievable BER floor level (for voice) ?<br>Note : BER floor level under BER measuring condition defined in Annex 2 using the data rates indicated in section 1 of Annex 2.<br>*See section 7* |
| ti | A1.3.1.2 | What is the achievable BER floor level (for data) ?<br>Note : BER floor level under BER measuring condition defined in Annex 2 using the data rates indicated in section 1 of Annex 2.<br>*See section 7* |
| ti | A1.3.1.3 | What is the maximum tolerable delay spread (in nsec) to maintain the voice and data service quality requirements ?<br>Note : The BER is an error floor level measured with the Doppler shift given in the BER measuring conditions of ANNEX 2.<br>*See section 7* |

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| ti | A1.3.1.4 | What is the maximum tolerable doppler shift (in Hz) to maintain the voice and data service quality requirements ?<br>Note : The BER is an error floor level measured with the delay spread given in the BER measuring conditions of ANNEX 2.<br>*See section 7* |
| ti | 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.<br>*See section 7.7* |
| ti | 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.<br>*See section 7.7.* |
| ti | 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.<br>*See section 7.7* |
| ti | 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.<br>*The SRTT supports sectorization.* |
| ti | 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.<br>*See annex 2.* |
| ti | 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.<br>*To be Completed.* |
| ti | 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.<br>*To be Completed.* |
| | A1.3.2 * | for satellite test environment only<br>*Not applicable to this phase of the study of the SRTT* |

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| ti | A1.3.2.1 * | What is the required C/No to achieve objective performance defined in ANNEX 2 ? |
| ti | A1.3.2.2 * | What are the Doppler compensation method and residual Doppler shift after compensation ? |
| ti | A1.3.2.3 * | Capacity : The spectrum efficiency of the radio transmission technology has to be evaluated assuming the deployment models described in ANNEX 2. |
| ti | A1.3.2.3. 1 * | What is the voice information capacity per required RF bandwidth (bits/sec/Hz) ? |
| ti | A1.3.2.3. 2 * | What is the voice plus data information capacity per required RF bandwidth (bits/sec/Hz) ? |
| ti | A1.3.2.4 * | Normalized power efficiency : The power efficiency of the radio transmission technology has to be evaluated assuming the deployment models described in ANNEX 2. |
| ti | A1.3.2.4. 1 * | What is the supported information bit rate per required carrier power-to-noise density ratio for the given channel performance under the given interference conditions for voice ? |
| ti | A1.3.2.4. 2 * | What is the 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 ? |
| ti | 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.<br>*See section 8* |
| ti | 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 ?<br>*Maximum distance is a function of the sensitivity and of the peak EIRP of the base station and of the mobile. They are not known yet. The only known limit is due to the guard period of the access burst which set the upper cell radius to 36 km.* |
| ti | A1.3.5 | Describe the capability for the use of repeaters<br>*To be Completed.* |
| ti | 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. :<br>- Does the SRTT 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.<br>*To be Completed.*<br>- Does the SRTT have the capability for the use of distributed antennas : Describe whether and how distributed antenna designs are used, and in which UMTS test environments.<br>*To be Completed.*<br>- Does the SRTT have the capability for the use of smart antennas (e.g. switched beam, adaptive, etc.) : Describe how smart antennas can be used and what is their impact on system performance.<br>*To be Completed.*<br>- Other antenna systems.<br>*To be Completed.* |

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| | A1.3.7 | Delay (for voice) |
| ti | 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.<br>*In general an exact value for the end-to-end delay depends on the frame duration of the speech codec, interleaving depth and other assumptions. Therefore such values should be carefully interpretted. As a preliminary statement, the W-TDMA concept allows the following delay values :*<br>*Speech Frame duration Interleaving depth Total delay Radio Transmission Delay*<br>*(Frames) (incl. Codec) (substracting Codec delay)*<br>*10 ms 2 23.075 ms 13.075 ms*<br>*20 ms 4 41.535 ms 21.535 ms*<br>*The first example provides for voice transmission with a low-delay reasonably comparable with current fixed networks, and with minimal echo cancellation performance requirements. The second example represents more typical operating conditions where additional interleaving is applied to improve robustness.*<br>*Processing delay has not been included in the above figures, since it will depend on the capabilities of the hardware. More detailed evaluation of this aspect is required.* |
| ti | 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.<br>*To be Completed.* |
| ti | A1.3.7.3 * | Does the proposed SRTT need echo control ? |
| ti | 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 G.711(64k PCM) and G.726 (32k ADPCM).<br>Note : If a special voice coding algorithm is indispensable for the proposed SRTT, the proponent should declare detail with its performance of the codec such as MOS level. (See A1.2.19) |
| ti | A1.3.9 | Description on the ability to sustain quality under certain extreme conditions.<br>*To be Completed.* |

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| ti | 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.<br>*To be Completed.* |
| ti | A1.3.9.2 | Hardware failures : Characterize system behavior and performance in such conditions. Provide detailed explanation on any calculation.<br>*To be Completed.* |
| ti | 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 ?<br>*Interference averaging is achieved by time and frequency hopping. Interference is avoided by DCA.* |
| ti | 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.<br>*Link adaptation maintains quality of service under different conditions.* |
| | A1.4 | Technology Design Constraints |
| ti | 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). |
| ti | A1.4.1.1 | For Base station transmission (terrestrial component only)<br>*To be Completed.* |
| ti | A1.4.1.2 | For Mobile station transmission<br>*To be Completed.* |
| ti | 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.<br>*To be Completed.* |

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| ti | A1.4.3 | Synchronisation requirements : Describe SRTT's timing requirements , e.g.<br>- Is base station-to-base station or satellite 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.<br>*Synchronisaton is not required at base station level when BCCH is permanently broadcast. In this environment, mobiles can be acquire the synchronisation of adjacent cells even within a call. Synchronisation might on the other end be required when non-constant BCCH is adopted.*<br>- Is base station-to-network synchronisation required ? (terrestrial only)<br>*BS synchronisation is not required, but improves spectrum efficiency..*<br>- State short-term frequency and timing accuracy of base station (or LES) transmit signal.<br>*To be Completed.*<br>- State source of external system reference and the accuracy required, if used at base station (or LES)(for example : derived from wireline network, or GPS receiver).<br>*Implementation dependant [to be confirmed].*<br>- State free run accuracy of mobile station frequency and timing reference clock.<br>*To be Completed.*<br>- State base-to-base bit time alignment requirement over a 24 hour period, in microseconds.<br>*To be Completed.*<br>- For private systems : can multiple unsynchronized systems coexist in the same environment ?<br>*Yes.* |
| ti | A1.4.4 | Timing jitter : For base (or LES) and mobile station give :<br>- the maximum jitter on the transmit signal,<br>*To be Completed.*<br>- the maximum jitter tolerated on the received signal.<br>*To be Completed.*<br>Timing jitter is defined as RMS value of the time variance normalized by symbol duration. |
| ti | A1.4.5 | Frequency synthesizer : What is the required step size, switched speed and frequency range of the frequency synthesizer of mobile stations ?<br>*The step size should be equal to the channel raster which is expected to be 200 KHz. The switched speed should correspond to the shortest possible burst (time for the transceiver to switch between transmit and receive state), that is to say 72 ms.* |
| ti | A1.4.6* | Does the proposed system require capabilities of fixed networks not generally available today ? |
| td | 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.<br>*See section 5.3.* |
| ti | A1.4.7 | Fixed network Feature Transparency |

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| ti | A1.4.7.1* | Which service(s) of the standard set of ISDN bearer services can the proposed SRTT pass to users without fixed network modification. |
| ti | 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 UMTS operating environment.<br>*To be Completed.* |
| ti | 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.<br>*To be Completed.* |
| ti | A1.4.10 | Characterize the linear and broadband transmitter requirements for base and mobile station. (terrestrial only)<br>*To be Completed.* |
| ti | A1.4.11 | Are linear receivers required ? Characterize the linearity requirements for the receivers for base and mobile station. (terrestrial only)<br>*To be Completed.* |
| ti | A1.4.12 | Specify the required dynamic range of receiver. (terrestrial only)<br>*To be Completed.* |

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| ti | A1.4.13 | What are the signal processing estimates for both the handportable and the base station ?<br>- MOPS (Mega Operation Per Second) value of parts processed by DSP<br>*Signal processing requirements of WB-TDMA equalizer.*<br>*The complexity of the 5-tap MLSE (SOVA=Soft Output Viterbi Equalizer) per detected symbol is about the same as the complexity of the current GSM equalizer. This SOVA equalizer as well as DFE (Decision Feedback Equalizer) has been used in the link level simulations. The number of real multiplications for SOVA are shown below:*<br>*Speech service, 1 slot (1/64)/frame, Bin-O-QAM, 5-tap MLSE, 4.8e6 real multiplicatios/second*<br>*144 kbit/s, 2 slots (1/16)/frame, Bin-O-QAM, 5-tap MLSE, 48e6 real multiplications/second*<br>*1 Mbit/s, 12 slots (1/16)/frame, Bin-O-QAM, 5-tap MLSE, 290e6 real multiplications/second*<br>*2 Mbit/s, 12 slots (1/16)/frame, Quat-O-QAM, 3-tap MLSE, 630e6 real multiplications/second*<br>*The corresponding figures for DFE are as follows:*<br>*Speech: 0.6e6 real multiplications/second*<br>*144 kbit/s: 6.0e6 real multiplications/second 1 Mbit/s: 36e6 real multiplications/second*<br>*2 Mbit/s: 28e6 real multiplications/second*<br>*The complexity of the DFE equalizer is much lower than the complexity of MLSE equalizer. This also implies that the remaining intersymbol interference with 5-tap MLSE equalizer could be cancelled with decision feedback part with only a minor increase in receiver complexity.*<br>- gate counts excluding DSP<br>*To be Completed.*<br>- ROM size requirements for DSP and gate counts in kByte<br>*To be Completed.*<br>- RAM size requirements for DSP and gate counts in kByte<br>*To be Completed.*<br>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.).<br>Note 2 : The signal processing estimates should be declared with the estimated condition such as assumed services, user bit rate and etc. |
| ti | A1.4.14* | Dropped calls : Describe how the SRTT handles dropped calls. Does the proposed SRTT utilize a transparent reconnect procedure - that is, the same as that employed for handoff ? |

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| ti | A1.4.15 | Characterize the frequency planning requirements :<br>- 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 ;<br>*Re-use 1 is supported*<br>- Characterize the frequency management between different cell layers ;<br>*To be Completed.*<br>- Does the SRTT use interleaved frequency allocation ?<br>*No.*<br>- Are there any frequency channels with particular planning requirements ?<br>*No.*<br>- Can the SRTT support self planning techniques ?<br>*Yes.*<br>- All other relevant requirements<br>Note : Interleaved frequency allocation is to allocate the $2^{nd}$ adjacent channel instead of adjacent channel at neighboring cluster cell. |
| ti | 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.<br>*The frame structure is compatible with the one of GSM, which makes the synchronisation of a MS to both systems feasible. As a result, a seamless inter-system handover is possible. An UMTS base station should maintain a list of GSM base station in the vicinity so that a MS operating in UMTS-mode can pre-synchronise with the strongest GSM base station in case an inter-system handover is required. The corresponding procedure can be applied in the reverse direction.* |
| ti | A1.4.16.1 | Does the SRTT support backwards compatibility into GSM/DCS in terms of easy dual mode terminal implementation, spectrum co-existence and handover between UMTS and GSM/DCS ?<br>*cf question A1.4.16* |
| ti | A1.4.17 | Are there any special requirements for base site implementation ? Are there any features which simplify implementation of base sites ? (terrestrial only)<br>*To be Completed.* |
| ti | 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.<br>*To be Completed.* |
| ti | A1.5.1 | What is the base station noise figure (dB) ?<br>*The actual figure is implementation dependant. The minimum requirement is TBC.* |
| ti | A1.5.2 | What is the mobile station noise figure (dB) ?<br>*The actual figure is implementation dependant. The minimum requirement is TBC.* |

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| ti | A1.5.3 | What is the base station antenna gain (dBi) ?<br>*This is implementation dependant. For sectorised envirnment, figures of 18 dBi can be considered.* |
| ti | A1.5.4 | What is the mobile station antenna gain (dBi) ?<br>*To be Completed.* |
| ti | A1.5.5 | What is the cable, connector and conbiner losses (dB) ?<br>*To be Completed.* |
| ti | A1.5.5 | What are the number of traffic channels per RF carrier ?<br>*To be completed* |
| ti | A1.5.6 | What is the SRTT operating point (BER/FER) for the required $E_b/N_0$ in the link budget template ?<br>*To be Completed.* |
| ti | A1.5.7 | What is the ratio of intra-sector interference to sum of intra-sector interference and inter-sector interference within a cell (dB) ?<br>*To be Completed.* |
| ti | A1.5.8 | What is the ratio of in-cell interference to total interference (dB) ?<br>*To be Completed.* |
| ti | A1.5.9 | What is the occupied bandwidth (99%) (Hz) ?<br>*To be Completed.* |
| ti | A1.5.10 | What is the information rate (dBHz) ?<br>*To be Completed.* |
| | A1.6 * | Satellite System Configuration (applicable to satellite component only) : Configuration details in this sub-section are not to be considered as variables. They are for information only.<br>*Not applicable to this SRTT.* |
| | A1.6.1 * | Configuration of satellite constellation |
| | A1.6.1.1 * | GSO, HEO, MEO, LEO or combination ? |
| | A1.6.1.2 * | What is the range of height where satellites are in active communication ? |
| | A1.6.1.3 * | What is the orbit inclination angle ? |
| | A1.6.1.4 * | What are the number of orbit planes ? |
| | A1.6.1.5 * | What are the number of satellites per orbit plane ? |
| | A1.6.2 * | What is the configuration of spot beams/cell layout pattern ? |
| | A1.6.3 * | What is the frequency reuse plan among spot beams ? |
| | A1.6.4 * | What is the service link G/T of satellite beam (average, minimum) ? |

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| | A1.6.5 * | What is the service link saturation EIRP of each beam (average, minimum), when configured to support 'Hot spot' ? |
| | A1.6.6 * | What is the service link total saturation EIRP per satellite ? |
| | A1.6.7 * | Satellite e.i.r.p. (effective isotropic radiated power) per RF carrier for satellite component |
| | A1.6.7.1 * | What is the maximum peak e.i.r.p. transmitted per RF carrier ? |
| | A1.6.7.2 * | What is the average e.i.r.p. transmitted per RF carrier ? |
| | A1.6.8 * | What is the feeder link information ? |
| | A1.6.9 * | What is the slot timing adjustment method (mainly applicable to TDMA system) ? |
| | A1.6.10 * | What is the satellite diversity method, if applicable ? |

# 11. Annex 2 : Operators Questions List

The question of operators following the SMG2 UMTS ad hoc meeting in Rennes, August 5-8 were reviewed and resulted in the following comments.

## *11.1 Guard band*

*1.1 What is the required guard band between two UMTS operators? Give assumptions made on transmission masks and minimum coupling loss factor to carry out your analysis.*

We consider two mobiles belonging to two different networks located in adjacent bands. We analyse how they can interfere first when their serving base stations are co-localised, then when they are not. This correspond to the following schemes :

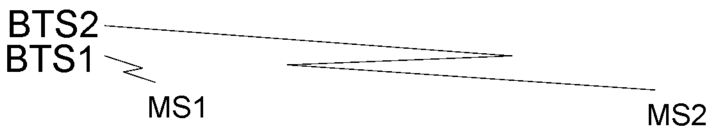


Figure 1 : Near-far effect for colocalised base stations

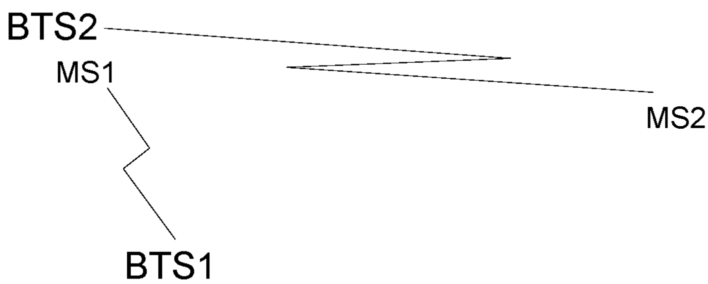


Figure 2 : Near-far effect for non-colocalised base stations

### 11.1.1 Principle of the analysis

We only consider the worst case for which MS1 is at the shortest possible distance considered and MS2 is at the cell edge. Both mobiles are in addition assumed to transmit during the same timeslot (or at least during overlapping timeslots : BTS synchronisation is not assumed and frames are *a priori* not aligned).

We indicate in the next two sub-sections the criterias allowing to determine whether a guard band is required or not. We define in the following section the considered environments and we derive the minimum path losses. The last section is a synthesis in the various environments presented. The lower and higher considered signal levels will be derived for the colocalised and non-colocalised cases, the opportunity of a guard band is in each case derived; the impact of time and frequency hopping is then mentioned.

The guardband requirements are derived from interference-limited scenarios. It must be pointed out that it leads to pessimistic results as the immunity brought by timeslot hopping, frequency hopping and link adaptation together with frequency reuse one and fractional loading[1] have not been considered in detail yet. This is for further study.

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[1] Interference averaging through frequency reuse one and fractional loading is described in section 5.2 of the WB-TDMA evaluation document version 1.0 c.

#### *11.1.1.1 Spurious Emissions of the mobiles*

The receiver of BTS2 can be affected by the spurious emissions of MS1 which can result from the power spectra of the modulation affected by the non-linearities of the power amplifier, the power ramping at the beginning and the end of the burst and the wideband noise of the local oscillator. The first effect is considered as dominant and is the only one considered in this preliminary study. According to initial simulations, the spurious level at the output of a conservative power amplifier is at about -25 dBc in the first adjacent channel and about -45 dBc in the second adjacent channel. It should be possible to improve these figures, this is left for further studies.

Simulations indicate that MS2 can be satisfactorily received from a C/(I+N) of roughly 7 dB (out of any margin). This can be achieved without guard band if :

- RX_Power(MS1, BTS2) - 25 dB < RX_Power(MS2, BTS2) - 7,
- where RX_Power(MS, BTS) is the power at which the MS is received by the BTS.

The power at which the interferer is received in the first adjacent channel must thus be at most 18 dB (25 - 7) above the power of the signal of the served mobile. Using the notation of GSM 05.05, we must have C/Ia1 of at least -18 dB.
(1)

A similar approach for the second adjacent channel result in a C/Ia2 of at least -38 dB. (2)

The guardband required between two un-coordinated UMTS systems is function of the C/Ia1 and C/Ia2 that can be provided in the considered environment.

#### *11.1.1.2 BTS receiver characteristics*

The receiver of the BTS can in addition be affected by a strong transmitter in an adjacent band due to the non-linearities of its LNA, the limited selectivity of its filters and to the wideband noise of its local oscillator causing a reciprocal mixing and resulting in some of the power in the adjacent bands to alias onto the useful signal.

We consider that the spurious emissions of the mobile in the adjacent bands are more constraining than the limitations of the receiver. Our analysis will thus focus on criteria (1) and (2) given in the previous section.

### 11.1.2 Considered environments and minimum coupling losses

The minimum coupling loss between MSs and the BTS depends on the environment. We consider below the models of path losses defined in annex 2 of 04.02, namely :

- Indoor office,
- Outdoor to indoor and Pedestrian,
- vehicular.

For simplification, we do not mix the environments in the analysis (it means that the path losses of both mobiles are assumed to be modelled by the same law).

We derive below the minimum and maximum coupling losses, from which we derive the highest possible power dynamic at the input of the base station, both for the colocated and non-colocated scenarios. These values are compared to the criteria (1) and (2) given above. The impact of time and frequency hopping is then considered.

The antenna gain of the mobile is in addition always 0 dBi and the minimum output power is assumed to be -10 dBm. The maximum output power is environment dependant; the assumed values is indicated in each case in section 3.

#### *11.1.2.1 Indoor Office*

We assume the minimum distance between a mobile and the base station antenna to be 1 metre in Indoor office. This results in a minimum path loss of 38.5 dB (we had to consider the free path loss as it gave a greater figure than the formula of section 1.4.1.1 of annex 2).

Considering a BTS antenna gain of 5 dBi and 2 dB of feeder loss, we derive a minimum coupling loss of 35.5 dB.

#### *11.1.2.2 Outdoor to Indoor and Pedestrian*

We assume the minimum distance between a mobile and the base station to be 6 metres. This results in a minimum path loss of 54 dB, considering again the free path loss propagation.

We now consider an antenna gain of 9 dBi and a feeder loss of 4 dB to derive a minimum coupling loss of 49 dB.

#### *11.1.2.3 Vehicular*

The antenna is assumed to be 25 metres above the ground level (following the assumption of annex 2 of 04.02, section 1.4.1.3 and assuming that the roof level is 10 metres above the ground). We assume that the minimum distance between the base station and the mobile is 100 metres (this is roughly the distance corresponding to the minimum coupling loss in line of sight considering the vertical beam pattern of the antenna). Under these assumptions, the minimum path loss is 90.5 dB.

Considering an antenna gain of 18 dBi and a feeder loss of 6 dB, we derive a minimum coupling loss of 78.5 dB.

### 11.1.3 Synthesis

We first determine the highest signal level received by the base station for each considered environment from the minimum coupling loss derived in section 2 and the assumed mobile transmit power.

The lowest admissible signal power is then stated for the considered environment by taking 10 dB over the BTS sensitivity for interference margin (cf. GSM rec. 05.50, Annex A) plus some shadowing margin (we provisionnaly take the C/I log-normal standard deviation assumed in annex 2 of 04.02 for the considered environment).

The figures obtain allow to see from criteria (1) and (2) whether a guard band is required.

#### *11.1.3.1 Colocated base stations*

We consider that the power control will work in a similar way for the two base stations : when the path loss between a mobile falls behind some threshold the serving base station orders the mobile to decrease its output power. As a consequence the mobile the closest from the base station, MS1 in our model, broadcasts with its smallest possible power which is -10 dBm with our assumptions.

Let determine the power dynamic range to determine the need of a guard band, considering that a correct demodulation of MS2 requires a C/I of at least 7 dB[2] :

[2] Operation at lower C/I is possible with link adaptation *ie* use of more transmission slots.

| Environment | Indoor Office | Indoor to Outdoor and Pedestrian | Vehicular |
|---|---|---|---|
| MS1 TX power | -10 dBm | | |
| Minimum coupling loss | 35.5 dB | 49 dB | 78.5 dB |
| MS1 RX power per BTS2 | -45.5 dBm | -59 dBm | -88.5 dBm |
| spurious level in 1st adjacent channel | -70.5 dBm | -84 dBm | -113.5 dBm |
| Required signal level from MS2 at BTS2 to avoid guard band. | **-63.5 dBm** | **-77 dBm** | **-106.5 dBm** |
| Spurious level in 2nd adjacent channel | -90.5 dBm | -104 dBm | -123.5 dBm |
| Required signal level from MS2 at BTS2 with one channel guardband (1.6 MHz). | **-83.5 dBm** | **-97 dBm** | (-126.5 dBm) |
| BTS reference sensitivity | -113 dBm | | |
| log-normal shadowing margin<br><br>interference margin | 12 dB<br><br>10 dB | 10 dB (outdoor)<br>12 dB (Indoor)<br>10 dB | 10 dB<br><br>10 dB |
| BTS sensitivity with interference and lognormal margins | **-91 dBm** | **-93 dBm (Outdoor)**<br>**-91 dBm (Indoor)** | **-93 dBm** |

It appears that in **vehicular environment** two UMTS systems can comfortably operate without guard band.

In Indoor to Outdoor environment and Indoor Office, a desensization might be observed without guard band. It is however possible to mitigate it through time and frequency hopping that should make very rare the conjunction of :

- the worst case shown on figure 1,
- the two mobiles transmitting on overlapping timeslots (although the probability of this event increases with the bit rate).

As a result **Indoor to Outdoor and Pedestrian** can be considered as being feasible without guard band. In the case that frequency hopping and timeslot hopping are not used, it is possible to adopt some guard band. We can see from the table above that a guard band of 1.6 MHz offers a protection which is superior by 4 to 6 dB to what is required. A guard band of 1 to 1.4 MHz should be sufficient to provide the required isolation.

A guard band of 1.6 MHz could be required for **Indoor Office**. Alternatively an indoor base station might be defined. It would be caracterised by an admissible input power range shifted by 20 to 25 dB as compared to a standard W-TDMA base station, following the approach that conducted to the definition of the microBTSs M1 to M3 for GSM.

### *11.1.3.2 Non-colocated base stations*

The reference situation is now described by Figure 2. The two base stations are now far away and MS1 can come in the vicinity of BTS2 while broadcasting at full power. We take for this peak transmit power the value indicated in the table given under section 3.3 of the "Response to the Operator's key questions to the UTRA concept groups", Tdoc SMG2/G23/97.

Let determine the power dynamic range to determine the need for a guard band in each of the considered environments.

| Environment | Indoor Office | Indoor to Outdoor and Pedestrian | Vehicular |
|---|---|---|---|
| MS1 TX power | 14 dBm | 24 dBm | 30 dBm |
| Minimum coupling loss | 35.5 dB | 49 dB | 78.5 dB |
| MS1 RX power per BTS2 | -21.5 dBm | -25 dBm | -48.5 dBm |
| Spurious level in first adjacent channel | -46.5 dBm | -50 dBm | -73.5 dBm |
| Required signal level from MS2 at BTS2 to avoid guard band | **-39.5 dBm** | **-43 dBm** | **-66.5 dBm** |
| Spurious level in second adjacent channel | -66.5 dBm | -70 dBm | -93.5 dBm |
| Required signal level from MS2 at BTS2 with one channel guardband (1.6 MHz) | **-59.5 dBm** | **-63 dBm** | **-86.5 dBm** |
| BTS reference sensitivity | -113 dBm | | |
| Lognormal shadowing margin<br><br>Interference margin | 12 dB<br><br>10 dB | 10 dB (outdoor)<br>12 dB (Indoor)<br>10 dB | 10 dB<br><br>10 dB |
| BTS sensitivity with interference and C/I margins | **-91 dBm** | **-93 dBm (Outdoor)**<br>**-91 dBm (Indoor)** | **-93 dBm** |

It appears that each environment meets some desensization, even with a guard band of 1.6 MHz. It can again be combatted by frequency and timeslot hopping.

The most critical case of Indoor office can again be delt with by the definition of a specific classes of indoor base station.

It must however be reminded that the probability of occurrence of scenarios conducting to these desensizations is very low as they only happen when :

- one interfering mobile is very close to the jammed base station,
- this mobile tranmits at full power,
- the timeslots during which it transmits are overlapping timeslots of a mobile received close to the BTS sensitivity.

*1.2 What is the required guard band between UMTS and other systems (in particular, with respect to GSM900, GSM1800, GSM1900, personal satellite communication systems, such as ICO, Iridium and Globalstar)?*

The required guard band between a W-TDMA and a GSM system located in adjacent bands is governed by the level of out-of-band emissions generated by each system.

W-TDMA presents some immunity against a GSM interferer because of its larger bandwidth. A guardband of probably 200, eventually 400 kHz might be required.

The guardband required to avoid serious disturbances caused by UMTS on GSM was not simulated yet. We can however anticipate (as a first approximation) from the analysis conducted in section 1.1 that the colocated scenario is much more confortable. It would correspond to the situation of a GSM operator starting to deploy an UMTS network from the site use from its GSM base stations.

## *11.2 Spectrum requirements*

*2.1 What is the minimum frequency bandwidth required for supporting 2 Mbit/s of user bit rate in each cell, indoor and low range outdoor, simultaneously, (i.e. an user bit rate of 2 Mbit/s/cell), in uplink and downlink respectively?*

*[What is the minimum required bandwidth for operating a network in an indoor and low range outdoor environment, each cell of which provides a circuit switched service of 2 Mbit/s user bit rate for one user in uplink and downlink respectively?]*

The required bandwidth depends on many things, however, two main issues are:

1. the amount of isolation between cells, on the other words, how interference limited the networks is
2. what kind of bearer distribution is assumed to generate the required 2 Mbits/s/cell

According to system simulations made so far, in a quite isolated environment, Outdoor to Indoor and Pedestrian with UDD384 bearers, the spectrum efficiency is 590 kb/s/MHz/cell, thus the required bandwidth for 2 Mb/s/cell is 3.4 MHz.

On the other hand,in the Indoor Office environment defined by 04.02 (hardly any isolation between cells, no walls and very high slow fading), the spectrum efficiency for UDD2048 is ca. 100 kb/s/MHz/cell. Thus the implied bandwidth for 2 Mb/s/cell would be 20 MHz. This reference environment is however not realistic in most of the cases and results in a too pessimistic spectrum efficiency. A more accurate approach would take walls and a more realistic attenuation factor into account.

The required bandwidth can in addition be reduced by introducing:

Modulation adaptation (see TDoc 15/97, pp 32-33)

Channel allocation to the cells (see TDoc 15/97, pp 42)

Quality based handover (see TDoc 15/97, pp 37-39)

In one cell case where all users are close to BTS, the required bandwidth is 0.67 MHz.

*2.2 What is the minimum required bandwidth and cluster size to deploy and operate a complete network providing a 2 Mbit/s real time service in the indoor environment and, in the same area and at the same time, a 384 kbit/s real time service in the pedestrian environment and a 144 kbit/s real time service in the vehicular environment (characterized by high mobility)? Please comment on the number of users this bandwidth would support at each rate/environment, and how the bandwidth increases when the number of users grows.*

To answer this important question, hierarchical cell structures (HCS) system simulations must be performed. The HCS simulations are for further study.

*2.3 How does the proposed UTRA concept support spectrum re-farming and what is the minimum bandwidth required?*

This question shall be answered altogether with question 1.2.

## *11.3 GSM backwards compatibility*

*3.1 Which concept for handover from UMTS to GSM and vice versa is used in the UTRA proposal? Which data services, besides speech, can be handed over between UMTS and GSM?*

Target is to provide full compatibility at radio interface between WB-TDMA and GSM, some adaptation are required if used service is not available in the target system. This could be handled prior or during the handover.

A description of the mechanisms allowing the handover between the two systems is given in the Evaluation Document (it appears in section 2.5.3 in the version 0.3 of this document).

*3.2 How will a UMTS/GSM dual mode terminal be implemented? What are the differences in terms of complexity and cost between a UMTS/GSM dual mode speech terminal and a GSM speech terminal?*

Nokia indicated that the difference of cost between a GSM speech terminal and a UMTS/GSM dual mode speech terminal was similar (difference of the order of 10%).

*3.3 Does the proposed UTRA concept have any limitations for the reuse of the GSM cell sites, e.g. limitations due to maximum cell size in an environment?*

As shown in the table below, the maximum range of WB-TDMA for speech service is longer than the range of GSM1800. So, cell site reuse is possible.

It should be noticed that the coverage analysis for speech service is based on 6 slot transmission. Those other 58 slots in a TDMA frame in that carrier can be used to support other users and this does not affect the range of the 6-slot user. For example, if other users take 1 slot/frame on average, 58 other users can be supported with one 1.6 MHz carrier while still providing the maximum range for one user. If this WB-TDMA scheme is compared to other proposals with e.g. 5 MHz bandwidth, then total of 3 WB-TDMA carriers can be used at the same bandwidth to support even a higher number of users while providing the maximum range.

It should also be noticed that the average transmission power in Vehicular environment is 4.3 dB lower that the maximum average power given in ETR0402. This is due to limitation set on the MS peak power. The lower average power also implies lower power consumption of the mobile station power amplifier at the cell edge.

| **WB-TDMA coverage analysis for speech service** | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| | | **WB-TDMA** | | **WB-TDMA** | | **WB-TDMA** | | **GSM1800** | | **GSM900** | |
| | | Downlink | Uplink | Downlink | Uplink | Downlink | Uplink | Downlink | Uplink | Downlink | Uplink |
| Test environment | | Indoor | Indoor | Pedestr. | Pedestr. | Vehicular | Vehicular | Vehicular | Vehicular | Vehicular | Vehicular |
| Multipath channel class | | A | A | A | A | A | A | | | | |
| Test service | | Speech | Speech | Speech | Speech | Speech | Speech | Speech | Speech | Speech | Speech |
| Number of slots used / frame | | 6 | 6 | 6 | 6 | 6 | 6 | 1 | 1 | 1 | 1 |
| Total number of slots / frame | | 64 | 64 | 64 | 64 | 64 | 64 | 8 | 8 | 8 | 8 |
| Bit rate | bit/s | 8000,00 | 8000,00 | 8000,00 | 8000,00 | 8000,00 | 8000,00 | 13000,00 | 13000,00 | 13000,00 | 13000,00 |
| Maximum peak power limitation | dBm | | 30,00 | | 30,00 | | 30,00 | | 30,00 | | 33,00 |
| Average TX power per traffic ch. (ETR0402 | dBm | 10,00 | 4,00 | 20,00 | 14,00 | 30,00 | 24,00 | 30,00 | 24,00 | 30,00 | 24,00 |
| Maximum TX power per traffic ch. | dBm | 20,28 | 14,28 | 30,28 | 24,28 | 40,28 | 30,00 | 39,03 | 30,00 | 39,03 | 33,00 |
| Average TX power per traffic ch. (real) | dBm | 10,00 | 4,00 | 20,00 | 14,00 | 30,00 | 19,72 | 30,00 | 20,97 | 30,00 | 23,97 |
| Maximum total TX power | dBm | 20,28 | 14,28 | 30,28 | 24,28 | 40,28 | 30,00 | 39,03 | 30,00 | 39,03 | 33,00 |
| Cable, conn. and combiner losses | dB | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 |
| TX antenna gain | dBi | 2,00 | 0,00 | 10,00 | 0,00 | 13,00 | 0,00 | 13,00 | 0,00 | 13,00 | 0,00 |
| TX EIRP per traffic channel | dBm | 20,28 | 14,28 | 38,28 | 24,28 | 51,28 | 30,00 | 50,03 | 30,00 | 50,03 | 33,00 |
| Total TX EIRP | dBm | 20,28 | 14,28 | 38,28 | 24,28 | 51,28 | 30,00 | 50,03 | 30,00 | 50,03 | 33,00 |
| RX antenna gain | dBi | 0,00 | 2,00 | 0,00 | 10,00 | 0,00 | 13,00 | 0,00 | 13,00 | 0,00 | 13,00 |
| Cable and connector losses | dB | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 |
| Receiver noise figure | dB | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 |
| Thermal noise density | dBm/Hz | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 |
| RX interference density | dBm/Hz | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 |
| Total effect. noise + interf. density | dBm/Hz | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 |
| Information rate (during tx) | dBHz | 49,31 | 49,31 | 49,31 | 49,31 | 49,31 | 49,31 | 50,17 | 50,17 | 50,17 | 50,17 |
| Required Eb/(No+Io) | dB | 8,40 | 5,90 | 8,60 | 5,90 | 8,70 | 6,60 | 12,00 | 12,00 | 12,00 | 12,00 |
| RX sensitivity | dB | -111,29 | -113,79 | -111,09 | -113,79 | -110,99 | -113,09 | -106,83 | -106,83 | -106,83 | -106,83 |
| Handoff gain | dB | 4,10 | 4,10 | 3,00 | 3,00 | 3,00 | 3,00 | 3,00 | 3,00 | 3,00 | 3,00 |
| Explicit diversity gain | dB | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 4,50 | 0,00 | 4,50 |
| Other gain | dB | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 |
| Log-normal fade margin | dB | 15,40 | 15,40 | 11,30 | 11,30 | 11,30 | 11,30 | 11,30 | 11,30 | 11,30 | 11,30 |
| Maximum path loss | dB | 120,27 | 116,77 | 141,07 | 137,77 | 153,97 | 145,79 | 148,56 | 144,03 | 148,56 | 147,03 |
| Maximum range | m | 596,54 | 456,01 | 669,85 | 553,96 | 4875,37 | 2954,25 | 3500,77 | 2652,54 | 5374,45 | 4893,49 |

*3.4 Is it possible to operate and manage the UTRA concept with a GSM operation and management system? If not, please indicate the expected differences.*

Due to backward compatibility WB-TDMA can in principle be operated through GSM O&M system. However, the goal for WB-TDMA has been easier network deployment e.g. with help of the Interference Averaging scheme (see document WB-TDMA Radio Resource Management ). Therefore such a heavy O&M system as in GSM is not expected to be required.

## *11.4 Macro diversity (or soft handover)*

*If macro diversity (or soft handover) is used in the proposed UTRA concept, please answer the following questions:*

Use of Macro diversity / soft HO is not required by the concept.

## *11.5 Support of asymmetric traffic*

*5.1 What is the concept for supporting asymmetric traffic between uplink and downlink and how do you propose to use the unpaired frequency bands?*

Asymmetric traffic can be supported by TDD or by allocating different service bit rates for uplink and downlink respectively in FDD. This can be done easily, as an example, by allocating a low bit rate carrier for the uplink from the “paired” band and a high bit rate carrier from the unpaired band. The unused pair of the low bit rate carrier can be used for asymmetric services within the symmetric band. TDMA makes this allocation scheme reasonable to implement.

TDD asymmetry could be achieved by 'traditional' means by allocation a fixed amount of slots through the whole operator's TDD spectrum for both directions.

However, due to the uncertainty concerning traffic profiles in different environments this approach does not seem to be very attractive. One idea in the usage of the TDD band is to allocate asymmetric capacity on an individual basis i.e. the frequency band would not be divided into purely uplink and downlink channels in the time domain but rather in a dynamic manner enabling the usage of all channels in frequency and time domain for both directions. These scheme is possible both by using RNC controlled Channel Allocation which enables fast handovers in the frequency and time domain or by Interference Averaging between different cells.

The described scheme would lead to minimum frequency planning. The operator does not have to put fixed percentage of the overall system capacity to up- or downlink traffic.

*5.2 If TDD mode is supported, what are the requirements for synchronization and how stringent are they? Would the use of TDD lower the range of the cell and by how much? Is TDD-FDD handover supported?*

Synchronisation is not required but optionally used. Synchronization results to some system performance gain. There are possibilities to use either over the air synchronisation mechanisms (even between uncordinated systems) or transmission based synchronisation within one system.

The dynamically allocated spectrum available to a particular operator could be based - in addition to other factors - on usage of syncronisation. This scheme would benefit operators who use syncronisation i.e. the ones who utilize the spectrum more effeiciently.

1) Synchronisation could improive spectrum efficiency of a system in a certain geographical area and the synchronisation issue is thus related to capacity of a system owned by a single operator.

2) The operator should have the option either to use or not to use synchronised BSSs in its designated frequency band ruled by the inter operator DCA scheme.

If TDD base stations are synchronized the accuracy of the synchronization should be better than the length of the guard period between bursts. i.e. $< 4$ µs, in order to have full advantage of the synchronization. It may also be possible to operate WB-TDMA / TDD network with asynchronos base stations by applying interference averaging to the interference between uplink and downlink. The asynchronous operation is for further study.

TDD provides the same range as FDD.

TDD-FDD handover is supported. This is considered very important in dualmode (FDD-TDD) usage enabling application transparency in both modes

*5.3 Does your concept support the use of 75% of an operator's total available spectrum in the downlink and 25% in the uplink? And the other way around (25% for the uplink and 75% for the uplink)? If so, please explain how it can be practically achieved and any implementation constraints that could be found.*

If the operators spectrum is unpaired (or spectrum pairs are considered separately), then the TDD option can dynamically adapt to any degree of traffic asymmetry. If the spectrum is paired (on an equal basis), then asymmetric FDD traffic will leave one of the band pairs under used. Additional capacity could be provided by operating in TDD mode in that part of the spectrum. Interference between TDD and FFD systems in the same band would be mitigated by the interference averaging concept, but this needs further study.

## *11.6 Operational requirements*

*6.1 Does the proposed UTRA concept support hierarchical cell structure (HCS) and if so how are single/multi-frequency handovers performed? If HCS is not used, how does the proposed UTRA concept deal with fast moving mobiles in micro cells?*

Hierachical Cell Structure are supported. Operation is similar to what is offered by GSM today.

*6.2 How does the proposed UTRA concept handle rapidly varying traffic distribution over short distances (as described in Tdoc SMG2 UMTS 70/97)?*

Rapidly varying traffic distribution can be handled by the MAC layer that can react quite rapidly. Interference averaging scheme also simplifies this management.

High traffic densities can be handled, in the limiting case by utilising all the radio resource in one cell. In this case transmission is likely to be noise limited rather than interference limited and the maximum traffic capacity will be correspondingly high (e.g. 1 to 3Mbps/MHz/Cell depending on the environment ).

*6.3 Are the data rates limited by the operator's willingness to build cell sites, or are there any other technical limitations? I.e., can an operator implement a high bit rate service wherever he wants to or are there any limitations that could make such a deployment technically impossible, e.g., power limitation in the mobile at very high data rates?*

Within the constraints of coverage and traffic capacity, bit rates are not constrained except by the available modulation schemes. In the anticipated deployment scenarios power limitations in the mobile are not expected to be a major practical constraint. Large delay spreads can be tolerated by definition of bursts with long enough training sequences.

If there is residual ISI after equalizer, it is handled by link adaptation and ARQ. Performance degradation is graceful as delay spread increases. Also Vehicular B channel can be supported with a small degradation in performance with the current burst structure. With longer training sequences and flexible burst structure the performance could be improved.

*6.4 Does your concept allow evolution to higher bit rates than those defined at present? If so, and assuming that spectrum is available, what is the limiting factor (other than range) in your concept to increase those rates? What changes would be required in the system?*

Highest possible bitrate is around 4 Mbit/s by allocating all the timeslots of a carrier to the same mobile.

*6.5 What is the maximum delay spread that can be handled in the test environments? Evaluate therefore the maximal bit rates that can be supported in each test environment and give the performance degradation as a function of the delay spread.*

The maximum delay spread that can be handled is linked to the size of the training sequence. The answer is similar to that of question 6.3 : long delay spreads can be adressed by flexible bursts

Maximum bit rates available with the bursts defined today are approximately:

Indoor, Outdoor to Indoor: more than 4 Mbit/s

Vehicular A: 2 Mbit/s

Vehicular B: 1 Mbit/s

*6.6 What strategy does your concept adopt to operate in different environments (e.g. business indoor, urban vehicular outdoor, rural outdoor, fixed outdoor)? Do any parameters need being changed depending on environment? If so, please specify which ones and an estimation of values for different environments, as well as how handover will be performed between them. Please consider the case of* *public and residential operators also as different environments and answer the above questions accordingly.*

To be completed.

*6.7 What is the system performances degradation due to implementation imperfections? In particular, what is the capacity loss due to inhomogeneous traffic distribution and none ideal cell site planning, e. g. cell site not at the traffic center, in comparison with the capacity for a homogenous traffic distribution and an ideal cell site planning?*

The interference averaging concept enables some possibilities to combat imperfect network planning: The slow DCA within the interference averaging concept implies that all carrier frequencies are available in *every* cell (i.e. 576 slots in case of 2x15 MHz). Then each cell has 64 most preferred slots, 64 second preferred slots, 64 third preferred slots, etc. If all cells activate only the 64 most preferred slots a reuse of 9 is obtained. If all cells activate all slots a reuse of 1 is obtained.

Probably a reuse of 1 is not the scenario that a real system would face, but a mixture of different reuses in different areas - depending on traffic load. So, when an imperfect planning has occurred, then some slots have to be de-activated in some cell(s) and activated in some other cell(s).

However, if there is no cell site at the hot spot this doesn't help...

The interference generated by the imperfect cell planning would in the interference averaging concept be averaged over the cells that use the same set of active slots.

Further, the MAC layer is also adaptive, for instance priority is given to delay-sensitive services.

Finally, it is however not possible to fully answer this question today. An exact figure of capacity loss would require simulations based on a defined model of the imperfection (user distribution, etc...).

*6.8 How does the proposed UTRA concept support uncoordinated operation of independent licensed (cellular, public) networks in adjacent frequency bands and within the same frequency band? How is the co-existence of licensed and unlicensed networks supported?*

Uncoordinated operation is possible both for FDD and TDD mode, with possibly a guard band in-between (this will be adressed with the answer to question 1.1).

FDD mode does not support spectrum sharing. This issue is for further study in TDD mode.

*6.9 If the UTRA concept requires synchronized base stations, what is the synchronisation accuracy requirement, and how will it be provided?*

BS synchronisation is not required, except for the bunch approach (cf. WB-TDMA Evaluation document, section 5.3.11 in version 0.3) , accuracy is in this last case of the order of guard period length.

## *11.7 Signalling overhead*

*7.1 What percentage overhead (such as power control, synchronization, handover procedures, support of asymmetric traffic etc.) is required for signalling (both circuit and packet switched services)?*

Proposed WB-TDMA UTRA concept is based on dynamic allocation of signalling resources, so all overhead caused by signalling is highly dependent on the cell load, traffic type, chosen RRM scheme etc. Almost all of the layer 2 information can be transmitted through one common set of logical channels (including both common and dedicated options) so the multiplexing gain is maximized, since the capacity for each message type does not to be separately estimated. Almost all RLC/MAC messages can be transmitted also without any overhead by stealing capacity from a suitable traffic bearer. Selection of the logical channel to be used can be done separately for each message.

In the following one possible example using only common control channels is considered. The percentage values are estimated based on the assumption that each cell would have allocatable capacity comparable to one complete 1.6 MHz WB-TDMA carrier. It is also assumed that the control signalling has the same modulation and coding overhead than the user data.

Power control

DL power is controlled by MS specific power control messages. Assuming the adjustment frequency to be 1 second the signalling requirement is 0,007% of UL capacity for each MS. Downlink powers can be controlled also separately for each bearer, but assuming on average one RT bearer for each MS the value is the same). Assuming 50 active MS's in each cell the resulting signalling overhead is 0,4 % from DL.

UL power is assumed to be either controlled similarily but twice as often as the DL power resulting into 0,014% of the DL capacity for each MS. (50 MSs require 0,8 % of DL capacity)

Alternatively UL power can be controlled separately for each slot frame by frame (200 Hz) which requires 1,5 % of DL capacity regardless of the amount of MS's.

Timing Advance

Average adjustment frequency for MS is assumed to be 0,5 s. This requires approximately 0,014% of DL capacity for each active MS. Assuming 50 MSs the resulting signalling overhead is 0,8 %)

L1, L2 and L3 BROADCAST INFORMATION

Synchronisation, pagings and other cell broadcasts require approximately 1,5 % of the primary DL carrier. (In case of multiple carriers in each cell the additonal carriers would have much smaller overhead)

Measurements

Amount of measurements for link quality is highly dependent on the chosen RRM scheme. E.g. the interference averaged concept does not require any link quality measurement signalling for link adaptation purposes. Neighbour cell measurements can also be transmitted over the air only on demand basis, i.e. when there is a need for a HO.

Real Time Services

If bitrate and link quality remain constant no additional control signalling is required for RT bearers. Real time traffic (Circuit switched services) require signalling capacity only for channel allocations/deallocations and for link adaptation. Assuming an average signalling interval of 200 ms:s the capacity requirement for each RT bearer is 0,03% of the DL and UL capacity. Assuming 50 RT bearers in each cell the resulting overhead is 1,5% from both UL and DL.

Non Real Time Services

Scheduled allocation procedure require constant signalling capacity regardless of the amount of active NRT connections.

Scheduled allocation procedure for DL data transfer requires appr. 1,5 % of the DL capacity and appr 1,5 % of the UL capacity. Uplink data transfer requires appr.1,5 % of the DL capacity and nothing from uplink.

One possible frame configuration:

Note! This is only one possible realisation of the logical channels. This is not by any means a minimum requirement for the protocol.

DL

· One 1/64 slot for L1, L2 and L3 broadcast information (incl pagings)

· One 1/64 slot for DL MAC messages (~200 messages/s)

Results to 3 % of DL capacity

UL

· One 1/64 slot for Random Access CHannels (~ 50 messages/s)

· One 1/64th slot for Uplink Acknowledgment channel (~200 messages/s)

Results to 3 % of UL capacity

If active Scheduled NRT transmission exists in DL additionally:

· DL: one 1/64th slot for Downlink NRT control (Enables maximum throughput)

· UL: one 1/64th slot for Ddownlink NRT control (The Forward Order Channel)

Results to 1,5 % of both UL and DL capacity

If active Scheduled NRTtransmission exists in UL additionally:

· DL: one 1/64th slot for Uplink NRT Control CHannel (Enables maximum throughput)

Results to 1,5 % of DL capacity

The complete set of control channels allowing all kind of traffic simultaneously results into 6 % of DL capacity and 4,5 % of uplink capacity.

*7.2 For packet switched services, does the proposed UTRA concept require transmission of control information (power control, timing advance information etc.) even when no packets need to be transmitted?*

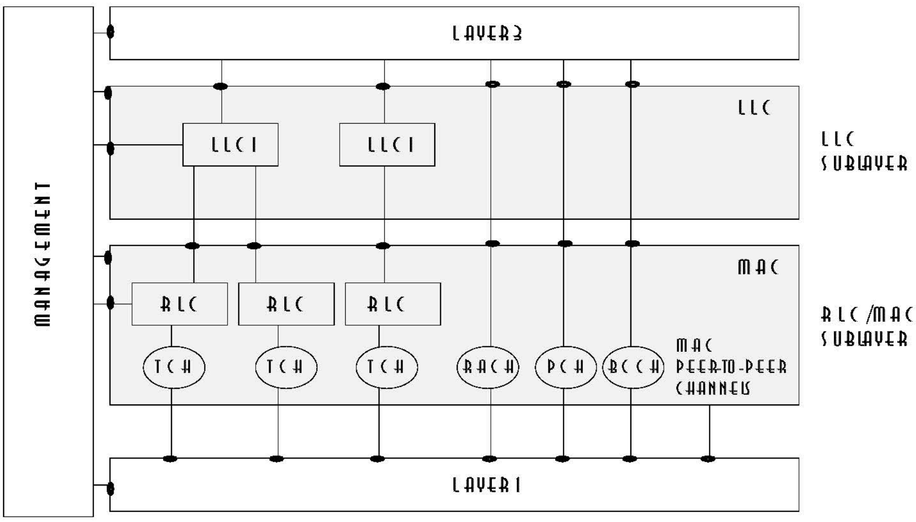
MSs having packet switced services require to have timing advance only when transmitting/receiving data. When no data is transmitted the timing advance signalling can be stopped. Nothing, however, prevents from maintaining the timing advance also when nothing is transmitted resulting into faster access to the radio resources at arrival of a new packet. (The access time saving is at least 10 milliseconds.)

Power control nor any other control messages related to 'packet protocol' are not required during the transmission breaks.

## 11.8 MAC (Medium Access Control) procedure

*81 What protocols are implemented for access contention, what is the strategy for resource allocation (slow and fast allocation, handling of mixed services, support of variable bit rate services and multimedia services), is there any functionality for adaptation to the radio environment or link quality, what protocols are implemented for packet switched services (including ARQ) ?*

RLC/MAC layer protocol provides fast resource allocations for real time (RT) and non real time (NRT) services supporting also variable bit rates and multibearer connections. For RT services QoS is fulfilled by means of dynamic link adaptation and for NRT services QoS can be maintained by effective ARQ.




The Radio Link Control/Medium Access Control (RLC/MAC) protocol supports two types of bearers, real time (RT) and non real time (NRT) bearers. The RT operation mode is used for the radio bearers which have strict delay constrains and quality is mainly fulfilled by power control and forward error corrections. The NRT operation mode is used for radio bearers with low delay requirements which allow backward error correction.

Functionalities of the RLC part and MAC part of the RLC/MAC protocol can be separated, even though both RLC and MAC have direct access to the physical layer and both parts provide services to the upper interface.

RLC entity is created in the MS and in the BS in association with each unidirectional radio bearer and its function is to guarantee the negotiated QoS for the radio bearer. A link adaptation algorithm within the RLC selects transmission format for the RT bearer service according to bearer QoS requirements and link quality. Transmission format is defined by coding, interleaving depth and modulation. On the transmitting side RLC handles the service data units (SDU) coming from the upper entity associated to the radio bearer. The RLC segments data into RLC protocol data units (RLC-PDU) according to the transmission format and forwards RLC-PDUs to the layer 1. On the receiving side RLC checks the CRC, if there is one, and possibly discards corrupted PDUs. Result of the CRC-check can be used in the ARQ procedure. RLC assembles the received PDUs and delivers the SDU to upper layer.

A channel allocation algorithm is located within the BSS side MAC entity. The MAC is responsible for the allocating, exchanging and releasing physical channels for the radio bearer. Peer-to-peer MAC messages between BSS side and mobile station are transmitted on the logical channels dedicated for MAC signalling. The MAC is not crossed by data flows coming from or destined to upper layers, that is the task of RLCs.

For RT radio bearers the MAC provides a resource allocation mechanism which allows a circuit switched type of reservation, i.e. the channel allocation is valid until the execution of a release procedure. For NRT radio bearers MAC provides a reservation mechanism, in which the reservation is valid only a certain allocation period. Fast allocation and release procedures for both RT and NRT bearers allow co-existence of these modes and guarantee efficient use of radio resources.

In the RT mode, the RLC entities request resources for the radio bearer due to radio condition variations and the bit rate variations. RLC resource requests are directed to MAC, which is responsible for the channel allocation signalling. Mobile initiated resource requests are transmitted on the dedicated control channel (DCCH) or random access channel (RACH). Channel allocations are transmitted on downlink DCCH or transmitted on the common control channel (CCCH). Fast associated control channel (FACCH) is a dedicated channel which uses capacity stolen from a bearer allocated to the MS. For a few occasional messages this is the preferred signalling channel. If FACCH can not be used, signalling can be transmitted on CCCH.

In the NRT mode, the RLC entities request resources for certain amount of data. For high bitrates 1/16 timeslot traffic channels are allocated for 9 ms allocation period (2 TDMA frames) at a time. Two frames gives some interleaving gain and is still very flexible. Channel allocations are announced on the NRT control channel (NCCH) and in order to avoid transmission of long identities there, a short reservation identity is allocated for the radio bearer. This identity is valid until the requested data is transmitted and during that time the mobile is obliged to listen to the NCCH. Traffic channels allocated for one reservation identity during one allocation period may vary from 0 to 14 timeslots, and the achieved bit rate may vary from 0 to 2 Mbit/s. For lower bitrates and infrequent transmissions the reservation is made from 1/16 or 1/64 timeslot traffic channels and reservation is valid for indicated time period.

The MAC is also responsible for ARQ signalling. CRC and reception quality based type II soft combining ARQ is expected to provide best efficiency.

8.2 Does your concept intend to treat circuit and packet oriented bearers as two separate entities or will a single type of bearer (e.g. packet) be used and both types of services be carried over the same bearer (e.g. support of connection oriented real time and delay constrained services over packet, similar to the way ATM operates)?

The same radio resources are shared by RT radio bearers and NRT radio bearers, but since they have fundamentally different QoS requirements they require different resource allocation procedures. For RT radio bearers the channel allocation is valid until execution of a release procedure (semi-circuit switched allocation). However, RT allocation and release procedures are very fast and they can be used to adapt resource assignments to RT variable bitrate data flows. For NRT radio bearers the reservation is valid only a certain allocation period (packet switched allocation). Short allocation period for NRT users guarantees that in the changing load conditions the resources can be reallocated to other users, e.g. RT users, after one allocation period.

See also 8.1.

## *11.9 System architecture requirements*

*9.1 What are the system architecture requirements of the proposed UTRA concept?*

System architecture is similar to that of GSM.

*9.2 If macro diversity is used, at which level are macro diversity combiners needed? Are transmission links between base station controllers required?*

Macro diversity is not assumed in the WB-TDMA proposal.

## *11.10 Radio network planning*

*10.1 What radio resource planning techniques are required? Is it necessary to plan handover in a UMTS network and how can it be planned?*

Purpose is to minimize the need for network planning. Thanks to Interference Averaging, network planning is not as sensitive as in GSM : non optimal planning should result in less degradation than in GSM.

*10.2 How can an operator do the coverage and capacity planning for a mixture of services?*

Coverage and capacity planning for a mixture of services is expected to needs as in GSM a network planning tool, but exact procedure is not known today.

*10.3 What solutions could be implemented to expand coverage and/or capacity? In particular, how does the UTRA concept support adaptive antenna?*

Solution to increase the capacity are described in the Evaluation Document (section Further Enhancements).

*10.4 Does coverage reduce with increasing traffic? If so indicate the relevant relationship and how to plan for coverage.*

There is no INTRA-CELL interference in a TDMA system, thus the coverage reduction is much smaller compared to the system with intra-cell interference. In case of a single cell, no coverage reduction is due to traffic load. See also answer of 3.3.

*10.5 Are there built-in functionalities to aid the monitoring and optimization of the radio interface performance and quality?*

Metrics can be made available to monitor and optimize the radio interface performance and quality, for instance amount of handovers, number of ARQ, feedback from link adaptation. Quality is guaranteed in realtime traffic by link adaptation and FEC and in non-realtime traffic with ARQ.

*10.6 Assuming 144 kbps coverage, what is the foreseen base station density relative to that of GSM 900 (full rate speech) in urban and rural environments?*

The range for 144 kbit/s of WB-TDMA in Vehicular channel is 2.1 km. The range for speech of GSM900 is 4.9 km. See the range calculation below. So, the range of WB-TDMA 144 kbit/s is 43 % of the range of GSM900 speech. Thus, the coverage area of WB-TDMA 144 kbit/s is 18 % of the coverage area of GSM900 speech. The base station density is therefore about 5.4 (=1/0.18) times higher than in GSM900 speech..

It must be pointed out however that base-station density requirements in GSM differ significantly for speech and data (particularly if requirement is BER=1.e-6).

## WB-TDMA coverage analysis for 144 kbit/s packet service

| | | WB-TDMA | | WB-TDMA | | WB-TDMA | | GSM1800 | | GSM900 | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| | | Downlink | Uplink | Downlink | Uplink | Downlink | Uplink | Downlink | Uplink | Downlink | Uplink |
| Test environment | | Indoor | Indoor | Pedestr. | Pedestr. | Vehicular | Vehicular | Vehicular | Vehicular | Vehicular | Vehicular |
| Multipath channel class | | A | A | A | A | A | A | | | | |
| Test service | | UDD144 | UDD144 | UDD144 | UDD144 | UDD144 | UDD144 | Speech | Speech | Speech | Speech |
| Number of slots used / frame | | 2 | 2 | 2 | 2 | 4 | 4 | 1 | 1 | 1 | 1 |
| Total number of slots / frame | | 16 | 16 | 16 | 16 | 16 | 16 | 8 | 8 | 8 | 8 |
| Bit rate | kbit/s | 144,00 | 144,00 | 144,00 | 144,00 | 144,00 | 144,00 | 13,00 | 13,00 | 13,00 | 13,00 |
| Maximum peak power limitation | dBm | | 30,00 | | 30,00 | | 30,00 | | 30,00 | | 33,00 |
| Average TX power per traffic ch. (ETR0402) | dBm | 10,00 | 4,00 | 20,00 | 14,00 | 30,00 | 24,00 | 30,00 | 24,00 | 30,00 | 24,00 |
| Maximum TX power per traffic ch. | dBm | 19,03 | 13,03 | 29,03 | 23,03 | 36,02 | 30,00 | 39,03 | 30,00 | 39,03 | 33,00 |
| Average TX power per traffic ch. (real) | dBm | 10,00 | 4,00 | 20,00 | 14,00 | 30,00 | 23,98 | 30,00 | 20,97 | 30,00 | 23,97 |
| Maximum total TX power | dBm | 19,03 | 13,03 | 29,03 | 23,03 | 36,02 | 30,00 | 39,03 | 30,00 | 39,03 | 33,00 |
| Cable, conn. and combiner losses | dB | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 |
| TX antenna gain | dBi | 2,00 | 0,00 | 10,00 | 0,00 | 13,00 | 0,00 | 13,00 | 0,00 | 13,00 | 0,00 |
| TX EIRP per traffic channel | dBm | 19,03 | 13,03 | 37,03 | 23,03 | 47,02 | 30,00 | 50,03 | 30,00 | 50,03 | 33,00 |
| Total TX EIRP | dBm | 19,03 | 13,03 | 37,03 | 23,03 | 47,02 | 30,00 | 50,03 | 30,00 | 50,03 | 33,00 |
| RX antenna gain | dBi | 0,00 | 2,00 | 0,00 | 10,00 | 0,00 | 13,00 | 0,00 | 13,00 | 0,00 | 13,00 |
| Cable and connector losses | dB | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 | 0,00 | 2,00 |
| Receiver noise figure | dB | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 | 5,00 |
| Thermal noise density | dBm/Hz | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 | -174,00 |
| RX interference density | dBm/Hz | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 | -1000,00 |
| Total effect. noise + interf. density | dBm/Hz | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 | -169,00 |
| Information rate (during tx) | dBHz | 60,61 | 60,61 | 60,61 | 60,61 | 57,60 | 57,60 | 50,17 | 50,17 | 50,17 | 50,17 |
| Required Eb/(No+Io) (taken from C/I results) | dB | 6,30 | 2,70 | 6,20 | 2,80 | 8,10 | 4,00 | 12,00 | 12,00 | 12,00 | 12,00 |
| RX sensitivity | dB | -102,09 | -105,69 | -102,19 | -105,59 | -103,30 | -107,40 | -106,83 | -106,83 | -106,83 | -106,83 |
| Handoff gain | dB | 4,10 | 4,10 | 3,00 | 3,00 | 3,00 | 3,00 | 3,00 | 3,00 | 3,00 | 3,00 |
| Explicit diversity gain | dB | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 4,50 | 0,00 | 4,50 |
| Other gain | dB | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 | 0,00 |
| Log-normal fade margin | dB | 15,40 | 15,40 | 11,30 | 11,30 | 11,30 | 11,30 | 11,30 | 11,30 | 11,30 | 11,30 |
| Maximum path loss | dB | 109,82 | 107,42 | 130,92 | 128,32 | 142,02 | 140,10 | 148,56 | 144,03 | 148,56 | 147,03 |
| Maximum range | m | 267,43 | 222,44 | 373,39 | 321,48 | 2344,86 | 2084,67 | 3500,77 | 2652,54 | 5374,45 | 4893,49 |

# 12. Annex 3 : Spectrum Efficiency Results Using an Analytical Model

## *12.1 Analytical Method*

### 12.1.1 Analysis Approach

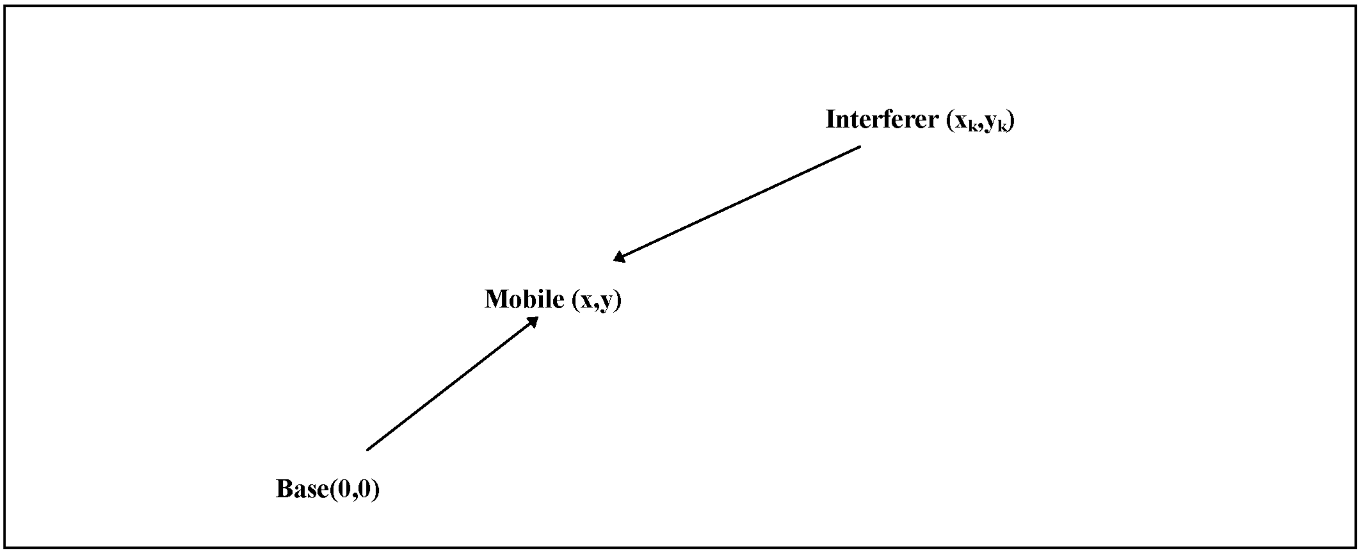
This section describes an analytical method for evaluating the performance of cellular mobile radio systems, in terms of spectral efficiency. It is intended that the results obtained using this method should be considered in support of more detailed simulation results presented elsewhere in the W-TDMA Evaluation Document

Link level simulation results (in the form of minimum carrier to interference ratio to meet BER performance requirements) are used to estimate the system capacity.

A number of simplifications have necessarily been introduced, including more analytically tractable deployment models then the simulation scenarios in UMTS 30.03. Therefore the results generated using this approach are probably best considered for illustrative purposes, and comparing options for UTRA, rather than in terms of absolute capacity.

#### *12.1.1.1 General Downlink Interference Model*

*Figure 6 Downlink Interference*




Here we consider the signal and interference powers in the downlink direction. In order to make the analysis viable, a number of simplifications and approximations are made. The interference limited case is studied, neglecting thermal noise. Adjacent channel interference is neglected.

The Mobile at position (x,y) receives a signal C(x,y) from the Base at (0,0) and interference, potentially from a number of sources I($x_k$,$y_k$), in this case other base stations. If the power transmitted by the wanted base station is P, and the path loss is L, with slow fading (Shadowing) $L_s$ and fast fading (Rayleigh) $L_f$, then the signal and the sum of interference powers can be computed from Equation 1.

*Equation 1*

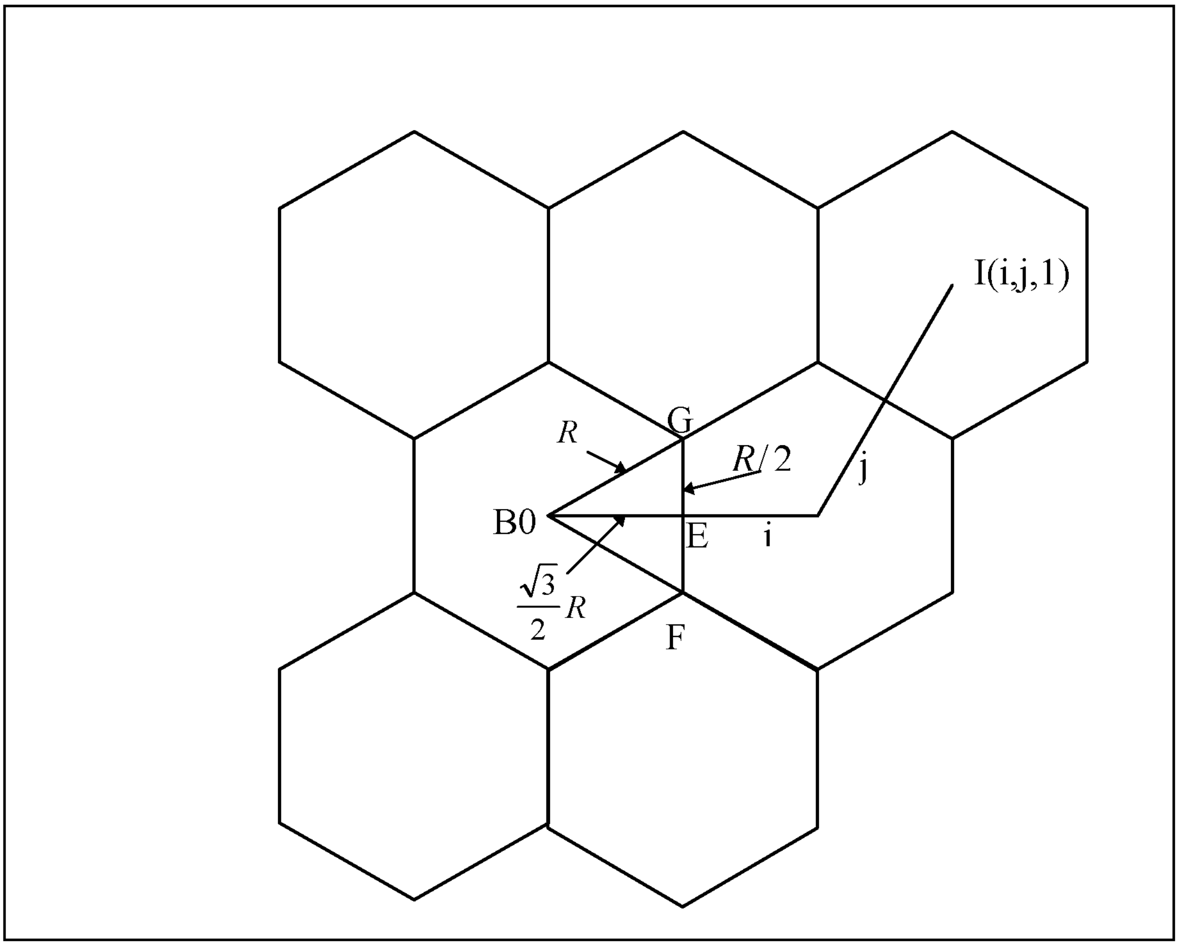
$$C(x,y) = \frac{P}{L_s L_f L(x,y)}$$

$$L(x,y) = \left(x^2 + y^2\right)^{\alpha/2}$$

$$I(x,y) = \sum_k \frac{P_k}{L_{sk} L_{fk}\left((x-x_k)^2 + (y-y_k)^2\right)^{\alpha/2}}$$

In general the locations of the interferers will depend on the cluster size. The cluster arrangement can be defined by parameters i,j such that the cluster size is given by $i^2+ij+j^2$. Then the location of interferers is as shown in *Figure 7*.

*Figure 7: Geometry of interferers*




The co-ordinates of the first eighteen interferers are given by Equation 2.

*Equation 2*

$$
\begin{array}{ll}
k = 1 & x(i,j,k) = i\sqrt{3}R + j\sqrt{3}R/2 \\
 & y(i,j,k) = j3R/2 \\
k = 2,..6 & x(i,j,k) = x(i,j,1)\cos((k-1)\pi/3) + y(i,j,1)\sin((k-1)\pi/3) \\
 & y(i,j,k) = y(i,j,1)\cos((k-1)\pi/3) - x(i,j,1)\sin((k-1)\pi/3) \\
k = 7 & x(i,j,k) = 2x(i,j,1) \\
 & y(i,j,k) = 2y(i,j,1) \\
k = 8...18 & x(i,j,k) = x(i,j,7)\cos((k-7)\pi/6) + y(i,j,7)\sin((k-7)\pi/6) \\
 & y(i,j,k) = y(i,j,7)\cos((k-7)\pi/6) - x(i,j,7)\sin((k-7)\pi/6)
\end{array}
$$

The first six interferers are distributed in a circle around the base station in question. The next twelve form a circle at twice that distance. In practice, the contribution from the second circle can often be neglected.

As an approximation we consider only the six nearest interferers (i.e. those in the first “ring”), and neglect the effect of fast fading (i.e. broad band transmission). Then signal and interference powers are given by Equation 3

*Equation 3*

$$C(x,y) = \frac{P}{L_s\left(x^2+y^2\right)^{\alpha/2}}$$

$$I(x,y) = \sum_{k=1}^{6} \frac{P_k}{L_s(k)\left(\left(x-(i,j,k)\right)^2 + \left(y-y(i,j,k)\right)^2\right)^{\alpha/2}}$$

The analysis is greatly simplified if we assume that all base stations continuously transmit the same power, such that $P=P_k$. This is reasonably valid for a fully loaded system without power control.

The usual model for $L_s$ is a log normal distributed random variable with zero mean and a standard deviation $\sigma_s$. As a further approximation we assume that I(x,y) is also log normally distributed with a mean value equal to the sum of the interference powers, and with the same variance. The signal to interference ratio (in dB) can then be written as in Equation 4.

*Equation 4*

$$SIR(x,y) = L_{eff} + 10\log_{10}\left(\frac{\sum_{k=1}^{6}\left(\left(x-x(i,j,k)\right)^2+\left(y-y(i,j,k)\right)^2\right)^{\alpha/2}}{\left(x^2+y^2\right)^{\alpha/2}}\right)$$

Here the variable $L_{eff}$ is the effective slow fading (shadowing) distribution. If the shadowing fading is uncorrelated for signals from the different base stations to the mobile, the effective variance is doubled and the standard deviation of $L_{eff}$ will be $\sqrt{2}\sigma_s$ (dB), with a mean of zero.

We can now compute the signal to interference ratio for a mobile at any location. However, because of the symmetry of the cell layout we only need to consider the triangular region B0 F G in Figure 7.

If $T_{SIR}$ is the minimum required signal to interference ratio for a communication channel, then the probability of achieving coverage is the probability that SIR is higher than this value. Equivalently the probability of an Outage is also given by the average probability that the SIR is less than some threshold $T_{SIR}$ as in Equation 5.

*Equation 5*

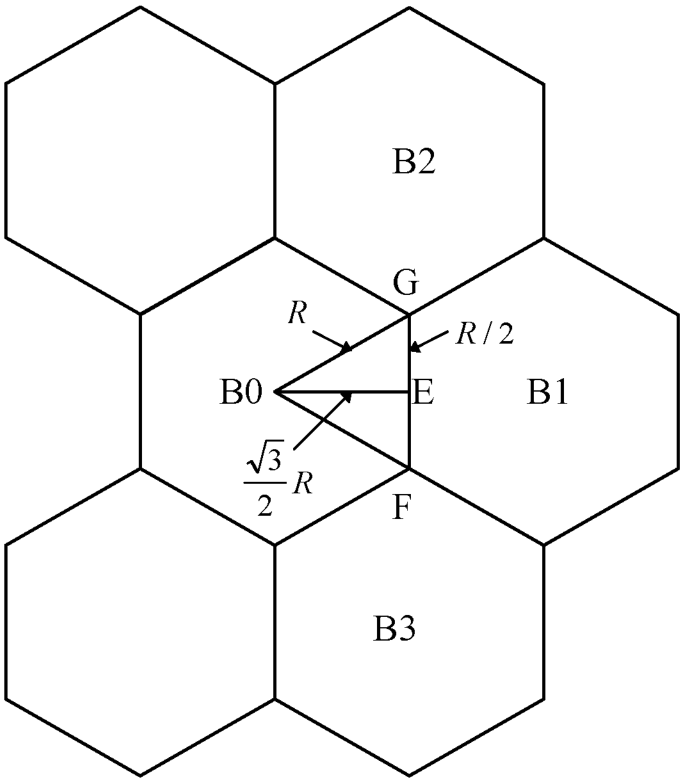
$$P_{out}(0) = \langle P(SIR < T_{SIR}) \rangle = \frac{\int_{x=0}^{x=R\sqrt{3}/2} \int_{y=-x/\sqrt{3}}^{y=x/\sqrt{3}} P(SIR(x,y) < T_{SIR})dydx}{\sqrt{3}R^2 / 4}$$

We note that $P(SIR(x,y)<T_{SIR}$ is obtained from the cumulative distribution of a Gaussian PDF with mean given by Equation 4 and standard deviation $\sqrt{2}\sigma_s$.

### 12.1.1.2 Handover to adjacent cells

Now we consider the possibility of handover for mobiles within the (area B0GF) to one of the three

*Figure 8: Geometry of handover to adjacent cells*




nearest neighbouring cells (B1,B2 or B3). In practice handovers could be made to other cells but it is assumed that this is a relatively unlikely event.

The probability of an outage with respect to a given base station (i.e. B1,B2 or B3) can be determine by substituting the co-ordinates (x,y) in Equation 5 with transformed co-ordinates (see Equation 6) relative to the selected base station. The resulting expression is given in Equation 7.

*Equation 6*

$$x_1 = x - R\sqrt{3}$$
$$y_1 = y$$
$$x_2 = x - R\sqrt{3}/2$$
$$y_2 = y - 3R/2$$
$$x_3 = x - R\sqrt{3}/2$$
$$y_3 = 3R/2 - y$$

*Equation 7*

$$P_{out}(n) = \langle P(SIR < T_{SIR}) \rangle = \frac{\int_{x=0}^{x=R\sqrt{3}/2} \int_{y=-x/\sqrt{3}}^{y=x/\sqrt{3}} P(SIR(x_n, y_n) < T_{SIR}) dydx}{\sqrt{3}R^2 / 4}$$

We assume that a handover is attempted if adequate SIR is not achieved with the current base station If we also assume that this can occur quickly enough to avoid loss of communication during the handover process, then the total outage probability is given by Equation 8.

*Equation 8*

$$P_{out\_total} = \prod_{n=0}^{N} P_{out}(n)$$

Where N is the total number of base-stations considered for handover.

The use of Equation 8 implies that the probabilities of handovers being possible to adjacent cells are independent, and also independent of the SIR for the current base station. This is a reasonable assumption for different carrier frequencies in the adjacent cells, but a relatively pessimistic assumption if any of the frequencies are the same. This occurs with cluster sizes 1 and 3. For cluster size 3 two of the adjacent cells will use the same frequency, but due to the geometrical arrangement a high probability of handover to one implies a lower probability to the other. Therefore Equation 8 remains a good approximation.

The case of adjacent cell re-use (with or without fractional loading) necessitates minor modification to the equations for handover. Firstly handover probabilities are now correlated. For example if the signal from the base in the current cell is weak, this increases the chance of a handover being possible to an adjacent cell, since the interference level for that cell is reduced. Therefore, when considering handover, the C/I ratio for the adjacent cells is computed without the contribution of an interference component from the current cell. In addition, handovers are only considered possible for cases where the required C/I ratio is less than 0dB.

### 12.1.2 Effect of Link Parameters

#### *12.1.2.1 Adaptive Modulation*

In general the principle of adaptive modulation requires that more than one modulation/channel coding scheme is a available (AM1, AM2....AMm). Each scheme will have an associated SIR threshold ($T_{SIR}$) for satisfactory operation (eg to meet BER requirement). At any given time the most bandwidth efficient option is selected, consistent with current SIR and BER requirement. This adaptation should happen quickly enough that the connection is not lost.

*Table 3 Adaptive Modulation with three options*

| **Modulation Scheme** | **Relative radio resource required** | **Required SIR** | **Outage probability without adaptation** | **Average fraction of time in cell coverage** | **Average fraction of radio resource used** |
|---|---|---|---|---|---|
| No coverage | | | | $P_1$ | |
| AM1 | $R_1$ | $T_{SIR1}$ | $P_1$ | $P_2$-$P_1$ | $R_1(P_2-P_1)/R$ |
| AM2 | $R_2$ | $T_{SIR2}$ | $P_2$ | $P_3$-$P_2$ | $R_2(P_3-P_2)/R$ |
| AM3 | $R_3$ | $T_{SIR3}$ | $P_3$ | 1-$P_3$ | $R_3(1-P_3)/R$ |

Note: $R = R_1(P_2-P_1)+R_2(P_3-P_2)+R_3(1-P_3)$

For a given cluster size, the value of $P_{out\_total}$ will increase monotonically with SIR threshold. Therefore the limiting outage probability for an adaptive modulation scheme is determined by the modulation with the lowest SIR threshold. Thus for three modulation schemes such that $T_{SIR1} < T_{SIR2} < T_{SIR3}$ , the fraction of time that each is used is given in Table 3

The fractional allocation of radio resource can be found from the resource required by each modulation scheme and the probability that it is used.

For the moment it is assumed that the coding/modulation scheme is selected according to C/I.

#### *12.1.2.2 Fractional Loading*

The effect of fractional loading is particularly relevant when considering frequency re-use in the adjacent cell.

With effective frequency/time hopping the effect of fractional loading can approximated by reducing the effective interference power by the loading factor.

This adjustment can be implemented in the analysis by reducing the SIR threshold $T_{SIR}$ by an appropriate amount. Usually this will not be large. For example with a typical loading level of 63%, and assuming a linear relation between loading and average interference power, the correction amounts to about 2dB.

#### *12.1.2.3 Power Control*

If link adaptation is used, this in general limits the power control dynamic range, since the C/I requirements of the different modulation/coding schemes typically differ by only a few dB. The effect of power control can be approximated by reducing the effective interference level (as for fractional loading). If the power distribution is uniform (for a given coding/modulation option) then the average interference reduction will be then typically be less than 3dB. This can be implemented in the analysis by reducing the SIR threshold $T_{SIR}$ by an appropriate amount.

For the moment we assume that power level is set according to C/I.

### 12.1.3 Blocking and Trunking Efficiency

Blocking must be considered as it is a major limiting factor on efficiency, and leads to low trunking efficiency when only a small number of communication channels are available. Traffic calls are assumed to originate uniformly within the cell and their arrival can be modelled by the Poisson probability distribution:

*Equation 9*

$$p_k = \frac{(\lambda t)^k}{k!}\exp(-\lambda t)$$

Here $p_k$ is the probability of k calls arriving in a time interval t, and $\lambda$ is the mean arrival rate of calls.

For the Poisson call model, if blocked calls are cleared (i.e. rejected), then the Erlang B formula for the fraction of blocked calls is appropriate.

*Equation 10*

$$E(A,N) = \frac{A^N / N!}{\sum_{i=0}^{N} A^i / i!}$$

Here A is the offered traffic in erlangs where $A = \lambda/\mu$ and $\mu$ is the mean call duration. N is the number of channels available.

We do not give an special consideration to handovers, under the assumption that handovers into the cell and out of it occur at equal rates.

In the case of adaptive modulation, we could consider blocking in the case of mixed traffic, where calls can require the use of different amounts of transmission capacity (i.e. different numbers of channels). This problem has been studied by Kaufman (IEEE Transactions on Communications, Vol COM-29, No 10, pp1474-1481, 1981).

A simpler approach, which will be valid for a large number of channels, is to assume that the traffic can be adequately represented by assuming that all calls use the average number of channels.

### 12.1.4 Successful Call Criteria

The criteria adopted for UMTS evaluation indicate that for a satisfied user the following conditions apply:

- The call should not be blocked on arrival.
- The error rate should be less than the required quality threshold ($10^{-3}$ or $10^{-6}$) for 95% of the call.
- The call should not be dropped.

For the moment we assume that the dropping criterion is much less stringent than the error rate criterion.

Blocking is considered in Section Blocking and Trunking Efficiency

Now, a call will have failed if the error rate exceeds the required threshold for more than the given fraction of the call (Q). Let us define the probability of the transmission being corrupted for a given (short) interval of time as $p_{corrupt}$. Then the probability of the call failing is the probability that the ratio of corrupted to uncorrupted intervals exceeds the required threshold. Considering a call of duration d time intervals, the failure probability is given by

*Equation 11*

$$p_{fail}(d,Q) = 1 - \sum_{k=0}^{d/Q} \frac{d}{(d-k)!\,k!} \, p_{corrupt}{}^{k} \left(1 - p_{corrupt}\right)^{d-k}$$

where d/Q is the maximum allowed number of corrupted time intervals during a call (which should be truncated to an integer value). In practice the calls have some duration distribution. In the case of a Poisson distribution with mean value D the average probability of failure is given by

*Equation 12*

$$p_{fail\_av}(D,Q) = \sum_{d=0}^{d=\infty} p_{fail}(d,Q) P(D,d) \,.$$

Where

$$P(D,d) = \frac{D^d e^{-D}}{d!}$$

In practice the summation is accurate enough with an upper limit of about 2D, and for the purposes of this paper, where the allowed number of corrupted durations is much greater than one, using the mean call duration in Equation 11 is a reasonable approximation. We note that for a time interval of 0.5 second (the resolution suggested in UMTS 30.03), an average call duration of 120secs gives D=240. For a quality threshold Q=0.05, an average call failure probability of 0.01 is achieved for a corruption probability of 0.0242 using Equation 12, and 0.0257 using the approximation of average call duration substituted in Equation 11.

This result can be interpreted as implying that more than 99% of calls will be satisfactory if the probability is less than about 0.02 of a short section of a call having an error rate greater than the allowed threshold.

### *12.1.4.1 Voice Calls*

Voice activity can be exploited to increase system capacity. If free resources can be re-allocated quickly enough, then it might be supposed that the effective number of available channels is increased in direct proportion to the inactivity factor. However, there will be a necessary signalling overhead, and on-going speech calls which become active and find insufficient resources available will become corrupted.

The following analysis for one carrier considers both to the case of free allocation of active speech calls to any transmission available slot(s), and to the case where speech calls are multiplexed together into a single burst. We also extend this to consider the case where a number of carriers are available, but a call remains assigned to the same carrier.

A voice call must remain sufficiently uncorrupted by "multiplex blocking" to remain acceptable. If the number of active calls exceeds the number of available slots, the information for some calls will be lost. If we assume that the affected calls are selected at random from the currently active calls, then with M channels available and N calls in progress (such that M<N), the average probability that a call will be corrupted during a short time interval is given by Equation 13.

*Equation 13*

$$p_{corrupt}(N,M) = \frac{1}{N}\sum_{i=M+1}^{N}(i-M)\frac{N!}{(N-i)!i!}p_a{}^{i}(1-p_a)^{N-i}$$

Where $p_a$ is the probability of the call being active.

The probability of the call failing can be derived by combining Equation 11 with Equation 13.

Now combining the probability that a call will be blocked on admission to the system (using Equation 10), and the probability that it will be lost due to insufficient slots (for any number of calls in progress), the average probability of losing a call is given approximately by Equation 14.

*Equation 14*

$$p_{loss}(A,N,M,N_{\max}) \approx E(A,N_{\max}) + \sum_{n=1}^{N_{\max}} p_{fail}(N,M)E(A,n)$$

where $N_{max}$ is the maximum number of calls allowed to be admitted at any one time.

We can further consider the case where more than one multiplex burst is used (eg on different carriers). In this case, if the calls are initially distributed as evenly as possible among the available carriers, then the average probability of call loss is derived from the probabilities of loss for each carrier.

*Equation 15*

$$p_{loss}(A,N,N_{\max},C) \approx E(A,N_{\max}) + \sum_{n=1}^{N_{\max}} E(A,n)\left(\frac{\sum_{c=1}^{C} Floor((n+c-1)/C)p_{fail}(Floor((n+c-1)/C),M_c)}{n}\right)$$

Where C is the number of carriers, $M_c$ is the number of channels available in the cth carrier. The function Floor(x) returns the largest integer less than or equal to x. Normally $M_c$ would have the same value for each carrier.

The analysis has not so far been extended to consider the case of mixed traffic (different slot allocations) in combination with voice activity, except using the approximation mentioned in Section Blocking and Trunking Efficiency.

### 12.1.5 System Capacity

#### *12.1.5.1 Stationary Terminals*

For stationary terminals, in the simplest case, the fraction of unsatisfied users is the sum of the probabilities of unsatisfactory coverage (using Equation 8) and blocking (using Equation 10). Thus the maximum traffic load can be found for which the number of satisfied users is sufficient.

For mixed traffic or where the calls can use different amounts of radio resource, approximate blocking results can be found by assuming that each call uses the average anoint of radio resource. For voice traffic with some activity factor, the probability of a satisfactory call is given by Equation 14 or Equation 15 for the multi-carrier case.

A slightly pessimistic approximation to the results from Equation 15 is obtained by considering the capacity available with a single carrier for 100% activity, dividing by the “inactivity factor” and multiplying by the total number of carriers.

In principle the SIR threshold should be adjusted in Equation 7 to account for fractional loading and power control. Then for typical conditions of 63% fractional loading, and 3dB power control adjustment, a total correction factor of zero will ensure that less than 1% of calls are dropped with a C/I measurement error of 2dB, and 0.1% with an error of 1.5dB.

#### *12.1.5.2 Moving terminals*

Further development of the above analysis is required for moving terminals.

If link adaptation and power control are fast enough to track changes in shadowing fading, then the main impact of terminal movement will be due to outages where the C/I falls bellow the minimum possible operating level. For fast moving terminals (or long enough calls) it is reasonable to assume that the C/I conditions for short segments of the call are statistically independent. This allows the use of the analysis in Section Successful Call Criteria. Therefore for a 1% probability of an unsatisfied user (due to BER), the outage probability for the most robust modulation/coding scheme must be less than about 0.025.

This would appear to give moving terminals a small advantage in call quality. However, three may be other practical factors which reduce this advantage (eg speed of link adaptation, Doppler effects, handover failure).

Therefore results presented here are for stationary users.

### 12.1.6 Procedure for Generation of Results

In a given radio environment the signal to interference ratio required to achieve an acceptable BER is obtained from Link Level simulations for different modulation and coding schemes.

Reasonable targets are to allow 1% loss of calls due insufficient quality or outage and 1% blocking, with a maximum allowed total of 2%.

In order to evaluate the capacity obtained for Adaptive Modulation the steps, carried out for each radio propagation model, are as follows:

- Select the set of modulation/coding schemes to be considered

- Obtain operating SIR for each modulation/coding scheme, and make probability of outage for each modulation/coding scheme for a range of cluster sizes.

- Set SIR threshold with corrections for power control and fractional loading if appropriate. Any correction for fractional loading may need to be reconsidered after the total capacity is determined.

- Select the smallest cluster size for which the required outage probability is obtained with the most robust available modulation scheme (i.e. outage probabilities of about 0.01 and 0.025 for stationary and moving terminals respectively).

- Obtain the maximum percentage of time that each modulation/coding scheme can be used, consistent with maintaining coverage.

- Compute the maximum possible blocking rate such that sufficient users remain satisfied (considering the achievable outage probability).

- Compute the maximum traffic loading at which the blocking rate criterion is met (using a mixed traffic model, with appropriate percentages of time and the radio resource required for a channel transmitted using each modulation scheme, or an approximation)

The analysis can be repeated with different modulation/coding schemes and possibly different cluster sizes. One assumption which may be necessary is that the multiple radio carriers are considered as a single resource.

## *12.2 GSM Spectrum Efficiency*

Comparison of the spectral efficiency of GSM with that of UMTS concepts seems to be of great interest in SMG2. However, such comparisons should be made using similar assumptions, bearer characteristics and quality of service. This document presents initial results for the downlink spectrum efficiency of GSM obtained for scenarios suitable for comparison with UMTS, and based on quality of service criteria compatible with those given in UMTS 30.03

The system is assumed to be interference limited, with all interfering base stations transmitting equal power. Downlink power control and fractional loading effects due to DTX are also considered. The model for stationary users is assumed.

### 12.2.1 GSM Link Level Performance

Performance requirements for GSM are given in pr ETS 300 577 (GSM 5.05 ver 4.19.0).

Since the UMTS bearers so far considered for evaluation all assume uniform error protection, this makes direct comparison with GSM speech services difficult. Therefore it is proposed that the comparison is based on the performance of the GSM data services.

The reference value for C/I is 9dB. For the TCH/F9.6 and H4.8 services, the maximum allowed BER under the reference conditions is as follows:

| Test Condition | BER |
|---|---|
| TU3, No FH | 8% |
| TU3, ideal FH | 0.3% |
| TU50, No FH | 0.8% |
| TU50, ideal FH | 0.3% |
| RA250, No FH | 0.2% |

The radio channels assumed for GSM are not very close to those to be used in the UMTS evaluation. However, there is reasonable similarity between the RA channel and the Outdoor to indoor channel A and Indoor channel B, and between the TU channel and Outdoor to indoor channel B and Vehicular channel A.

Based on the above it seems reasonable to suppose, as a starting assumption, that GSM can achieve a BER of 0.1% (required for speech) with C/I=10dB, and 0.0001% (required for data) with C/I=12dB (approx.), provided frequency hopping approaching the ideal is used. We can approximate the effect of DTX with 50% voice activity by reducing the effective C/I requirement by 3dB, since the average interference power is reduced by a factor of two. An additional reduction of 3dB is included for power control. This assumes sufficient interferer diversity is achieved by frequency hopping.

The TCH/F9.6 bit rate is close enough to the nominal 8kbps assumed for speech in UMTS to make such a comparison reasonable.

All the spectrum efficiency results given below are under the assumption of 9.6kbps throughput per channel.

### 12.2.2 Available GSM channels

In 15MHz bandwidth, as proposed for UTRA evaluation, there is room for up to 75 carriers with 200kHz spacing (assuming no guard bands). Each of these can support a maximum of 8 voice or 9.6kbps data calls. Therefore the maximum number of channels available is 600.

With cluster sizes larger than one the number of carriers per cell is reduced accordingly.

### 12.2.3 Deployment Results - Indoor Environment

In the indoor environment, the propagation exponent is 3, and the standard deviation of log-normal shadowing is 12dB. An un-sectored cell is appropriate.

Under these conditions the probability of outage for “Voice” (10dB C/I) and “Data” (12dB C/I) are given for some typical cluster sizes.

*Table 4 Outage probability in indoor environment*

| Cluster Size | Outage Probability - Voice | Outage Probability - Voice (DTX+Power Control) | Outage Probability - Data |
|---|---|---|---|
| 3 | 0.155 | 0.060 | 0.20 |
| 4 | 0.110 | 0.037 | 0.147 |
| 7 | 0.054 | 0.0148 | 0.077 |
| 9 | 0.038 | 0.0094 | 0.056 |
| 12 | 0.025 | 0.0055 | 0.038 |
| 13 | 0.022 | 0.0046 | 0.038 |
| 19 | 0.0117 | 0.0021 | 0.0190 |

It is clear that for voice with DTX and power control, a cluster size of 7 is sufficient to satisfy 98% of users. This gives 75/7=10 carriers per cell. A cluster size of 19 is needed for the other options. The spectrum efficiency can now be calculated under the maximum traffic condition where the sum of the blocking and outage probabilities is 0.02.

*Table 5: GSM Spectrum Efficiency in indoor environment*

| | Voice | Voice (DTX+Power Control) | Data (9.6kbs) |
|---|---|---|---|
| Cluster size | 19 | 7 | 19 |
| Carriers per cell | 3 | 10 | 3 |
| Channels per cell | 24 | 80 | 24 |
| Total spectrum used (MHz) | 11.4 | 14.0 | 11.4 |
| Outage probability | 0.0117 | 0.0148 | 0.0190 |
| Maximum blocking probability | 0.0083 | 0.0052 | 0.001 |
| Maximum traffic per cell (Erlangs) | 15.0 | 62.8 | 12.2 |
| Spectrum Efficiency (Erlang/MHz/cell) | 1.32 | 4.48 | 0.93 |
| Spectrum Efficiency (kbps/MHz/cell) | 12.7 | 21.5 | 8.9 |

### 12.2.4 Deployment Results - Outdoor environment

In the outdoor environment, the propagation exponent is 4 (depending on antenna configuration), and the standard deviation of log-normal shadowing is 10dB. The results here are given for omni-directional antennas.

Under these conditions the probability of outage for “Voice” (10dB C/I) and “Data” (12dB C/I) is given for some typical cluster sizes.

*Table 6: Outage Probability in outdoor environment*

| Cluster Size | Outage Probability - Voice | Outage Probability - Voice (DTX+Power Control) | Outage Probability - Data |
|---|---|---|---|
| 3 | 0.087 | 0.023 | 0.123 |
| 4 | 0.044 | 0.0091 | 0.068 |
| 7 | 0.0101 | 0.00130 | 0.0180 |
| 9 | 0.0047 | 0.00048 | 0.0088 |
| 12 | 0.00178 | 0.00054 | 0.0037 |
| 13 | 0.00135 | 0.000099 | 0.0028 |
| 19 | 0.00031 | 0.0000166 | 0.00073 |

In the case of Voice with DTX and power control, a cluster size of 4 is needed to meet the requirement that 98% of users must be satisfied. This means that there would be 75/4 = 18 carriers available per cell, giving 144 voice channels per cell. In the other cases the cluster size is 7, giving 80 channels per cell.

The spectrum efficiency can now be calculated

*Table 7: GSM Spectrum Efficiency in outdoor environment*

| | Voice | Voice (DTX+Power Control) | Data (9.6kbs) |
|---|---|---|---|
| Cluster size | 7 | 4 | 7 |
| Carriers per cell | 10 | 18 | 10 |
| Channels per cell | 80 | 144 | 80 |
| Total spectrum used (MHz) | 14 | 14.4 | 14 |
| Outage probability | 0.0101 | 0.0091 | 0.018 |
| Maximum blocking probability | 0.0099 | 0.0109 | 0.002 |
| Maximum traffic per cell (Erlangs) | 65 | 126 | 60 |
| Spectrum Efficiency (Erlang/MHz/cell) | 4.6 | 8.8 | 4.3 |
| Spectrum Efficiency (kbps/MHz/cell) | 44 | 42 | 41 |

For sectored cells the efficiency should be similar to the above values.

### 12.2.5 Conclusions

Making some reasonable assumptions the spectrum efficiency has been obtained for GSM in a way which allows comparisons to be made with the voice bearer currently being evaluated for UTRA. Spectrum efficiency has also been calculated for the GSM data services using a quality criterion which is similar to that used in UMTS evaluation.

The results presented here clearly show that:

- The radio environment, and particularly the propagation model, has a major effect on spectral efficiency (typically by a factor of more than two between indoor and outdoor environments).

- Minimising the cluster size while maintaining low enough outage probability is essential in maximising spectrum efficiency.

- Any measures to reduce interference levels such as DTX, fractional loading and power control will significantly improve spectral efficiency (at least in terms of Erlangs).

It is interesting to compare the results here with those produced by Wigard et al (Capacity of a GSM Network with Fractional Loading and Random Frequency Hopping, 7th IEEE International Symposium on Personal, Indoor and Mobile Radio Communications, p723-72, Taiwan, October 1996). Although some assumptions were rather different (eg propagation exponent of 3.5, and lower quality thresholds), the capacity for cluster size of 4 was about 8.3 Erlang/MHz/Cell compared with the 8.8 Erlang/MHz/Cell here.

These results are also similar to those for the GSM reference model proposed for comparison with UMTS (Tdoc SMG2 UMTS 122/97) estimates the efficiency as 2.35 Erlang/MHz/Cell and 30.55kbps/MHz/Cell, but with sectored cells and without frequency hopping or DTX.

### *12.3 Spectrum Efficiency of W-TDMA*

Results are presented here for the downlink spectrum efficiency of the W-TDMA concept for various deployment options. The model for stationary users is adopted, but similar results would be obtained assuming moving terminals. It is assumed that there is a 3dB improvement in average C/I from the use of power control, and an additional 2dB improvement for fractional loading (nominally of 63%) in the case of adjacent cell frequency re-use.

Results are given for omni-directional antennas and hexagonal cell layout. Cluster sizes of 1 and 3 are considered.

In 15MHz bandwidth up to nine 1.6MHz carriers are available (occupying a total bandwidth of 14.4MHz). It is assumed that new calls can be allocated to any available radio resources. In the case of speech calls with 50% activity factor it is assumed that a call remains on the initially allocated carrier. This may be pessimistic, since inter-frequency handovers are allowed. In general it is assumed for data calls that radio resources are effectively pooled (i.e. a call can transferred to a new carrier as required to optimise slot allocations).

Admission control is assumed to operate such that the call is blocked if insufficient resources are available in that cell. The possibility of serving the call from a nearby cell is not considered. This seems a pessimistic assumption

Signalling capacity equivalent to 1/16 of a frame on every carrier is assumed to be fully occupied (eg for link adaptation and power control), and this included in the spectrum efficiency calculation.

#### 12.3.1 Deployment Results - Pedestrian Environment

In the Pedestrian Environment the propagation exponent is 4, and the standard deviation of log-normal shadowing is 10dB. Simulations carried out for the Outdoor to Indoor case, with a standard deviation of shadowing of 12dB gave similar results to those below.

##### *12.3.1.1 Speech*

In this case 64 slots per frame are available. It is assumed that 4 are used for signalling. Single frequency re-use can be supported. Fractional loading and power control are assumed, and the fraction of the time each modulation/coding scheme is used is shown.

*Table 8: Modulation/coding schemes (Speech, Pedestrian A)*

| Number of slots per frame | Modulation | C/I (dB) | Fraction of time used |
|---|---|---|---|
| 3 | B-OQAM | -1.4 | 0.237 |
| 2 | B-OQAM | 2.8 | 0.103 |
| 1 | B-OQAM | 7.6 | 0.248 |
| 1 in 2 frames | B-OQAM | 18.2 | 0.398 |

The average number of slots per user channel is 1.36. The effective number of channels per carrier is therefore 43. Coverage with sufficient quality of service is achieved for 99.4% of users. Allowing a blocking probability of 0.006 this gives a total capacity of 356 Erlangs/Cell. Considering the extra slots available with 50% voice activity, this increases to 546 Erlangs/Cell, equivalent to 71% fractional loading. This gives 152kbps/MHz/Cell.

These results may be pessimistic, since the blocking and bad quality calculation is carried out per carrier. In practice new calls can be allocated to any carrier with unused slots, increasing the trunking efficiency.

##### *12.3.1.2 LCD 144*

In this case there are 16 slots per carrier, with an average of one slot allocated to signalling. Two cluster sizes are considered.

*Table 9: Modulation/coding schemes (LCD 144, Pedestrian A)*

| Number of slots per frame | Modulation | C/I (dB) | Fraction of time used (Schema A, re-use=1) | Fraction of time used (Scheme B, re-use=3) |
|---|---|---|---|---|
| 8 | B-OQAM | -2.0 (est) | 0.205 | - |
| 4 | B-OQAM | 1.0 | 0.059 | - |
| 3 | B-OQAM | 4.0 | 0.134 | 0.038 |
| 2 | Q-OQAM | 10.0 | 0.275 | 0.262 |
| 1 | Q-OQAM | 22.0 | 0.316 | 0.690 |

With re-use pattern 3 the average number of slots per channel is 1.33, which gives 11 effective channels per carrier, and a capacity of 23 Erlangs per cell or 230kbps/MHz/Cell with a fractional loading of 70%. With adjacent cell re-use the average number of slots is 3.1, which gives 4 effective channels per carrier, and a capacity of 25 Erlangs per cell or 250kbps/MHz/Cell with fractional loading of 60%.

These figures may be optimistic since the number of effective channels per carrier is small, which may lead to significant additional blocking.

With re-use factor 1, it appears is necessary to use a larger number of slots (eg 8) per channel to achieve sufficient coverage. Smaller slot allocations are sufficient for re-use factor 3.

## 12.3.2 Deployment Results - Vehicular Environment

In UMTS 30.03 the vehicular environment specifies sectored cells. The results here are presented for omni-directional antennas. The capacity figures per cell (i.e. per sector) should be similar for sectored cells. The standard deviation of shadowing was 10dB, with a propagation exponent assumed to be 4.

### *12.3.2.1 Speech*

Here we consider as examples two possible schemes;

A: Single frequency re-use with link adaptation options of 1, 2, 4 and 6 slots per channel

B: Cluster size of 3 with 1/4, 1/2. 1 and 2 slot option.

For scheme A the C/I the offset for power control and fractional loading is 5dB. For scheme B only a 3dB offset for power control is applied. In scheme B the 1/2 and 1/4 slot options require the use of a multiplexed burst.

*Table 10: Modulation/coding schemes (Speech, Vehicular A)*

| Number of slots per frame | Modulation | C/I (dB) | Fraction of time used (Scheme A, re-use=1) | Fraction of time used (Scheme B, re-use=3) |
|---|---|---|---|---|
| 6 | B-OQAM | -4.1 | 0.006 | - |
| 4 | B-OQAM | -1.9 | 0.227 | - |
| 2 | B-OQAM | 2.2 | 0.121 | 0.022 |
| 1 | B-OQAM | 7.7 | 0.644 | 0.056 |
| 1/2 | B-OQAM | 12.8 | - | 0.258 |
| 1/4 | Q-OQAM | 23 (est) | - | 0.658 |

***For comparison we also consider Scheme A in the Vehicular B channel.***

*Table 11: Modulation/coding schemes (Speech, Vehicular B)*

| Number of slots per frame | Modulation | C/I (dB) | Fraction of time used (Scheme A, re-use=1) |
|---|---|---|---|
| 6 | B-OQAM | --3.2 | 0.008 |
| 4 | B-OQAM | -0.9 | 0.263 |
| 2 | B-OQAM | 4.2 | 0.202 |
| 1 | B-OQAM | 13.0 | 0.519 |

*Table 12 Spectral Efficiency (Speech, Vehicular A)*

| Scheme | Cluster size | Average number of slots per channel | Spectrum Efficiency (Erlang/Cell) | Spectrum Efficiency (kbps/MHz/cell) | Fractional loading (%) |
|---|---|---|---|---|---|
| A (Veh A) | 1 | 1.83 | 411 | 114 | 72 |
| A (Veh B) | 1 | 2.02 | 357 | 99 | 69 |
| B | 3 | 0.39 | 815 | 226 | 89 |

If can be seen that with Scheme A in re-use factor 1 the spectral efficiency is reduced by about 13% in the Vehicular B environment. This reduction might be larger with Scheme B, since shorter slot allocations would typically be used.

### *12.3.2.2 LCD384*

In this case we consider 16 slots in a frame, and a 384kbps circuit switched bearer with a 3 frequency re-use pattern.

*Table 13: Modulation/coding schemes (LCD 384, Vehicular A)*

| Number of slots per frame | Modulation | C/I (dB) | Fraction of time used (Cluster size=3) |
|---|---|---|---|
| 8 | B-OQAM | 3.0 | 0.015 |
| 6 | B-OQAM | 7.0 | 0.100 |
| 4 | Q-OQAM | 15.0 | 0.097 |

| | | | |
|---|---|---|---|
| 3 | Q-OQAM | 19 (est) | 0.780 |

The average number of slots per channel is 3.4, giving 4 channels per carrier. The spectrum efficiency is 6.1 Erlangs/Cell or 163kbps/MHz/Cell. The average fractional loading is 49%. Clearly the small number of available channels limits the trunking efficiency.

In this case the use of even shorter slot allocations appears advantageous.

Single frequency re-use would probably be viable (eg by introducing a 12 or 16 slot mode), but at a low fractional loading level.

## 12.3.3 Deployment Results - Indoor Environment

The deployment model in UMTS 30.03 specifies a rectangular geometry for the base-stations. Here we consider a conventional hexagonal deployment. This is expected to give somewhat higher spectral efficiency. The standard deviation of shadowing is 12dB with a propagation exponent of 3.

### *12.3.3.1 Speech*

We consider two options with re-use factor 1.

*Table 14 Modulation/coding schemes (Speech, Indoor A)*

| Number of slots per frame | Modulation | C/I (dB) | Fraction of time used (Scheme A re-use 1) | Fraction of time used (Scheme B re-use 1) |
|---|---|---|---|---|
| 6 | B-OQAM | -5.1 | 0.008 | 0.008 |
| 4 | B-OQAM | -2.8 | 0.009 | 0.009 |
| 3 | B-OQAM | -1.0 | 0.290 | 0.290 |
| 2 | B-OQAM | 1.7 | 0.121 | 0.121 |
| 1 | B-OQAM | 7.5 | 0.246 | 0.128 |
| 1 in 2 frames | B-OQAM | 19.2 | 0.316 | - |
| 1/2 | B-OQAM | 13.4 | | 0.203 |
| 1/4 | Q-OQAM | 24 (est) | | 0.231 |

*Table 15 Spectrum Efficiency (Speech, Indoor A)*

| Scheme | Cluster size | Average number of slots per channel | Spectrum Efficiency (Erlang/Cell) | Spectrum Efficiency (kbps/MHz/cell) | Fractional loading (%) |
|---|---|---|---|---|---|
| A | 1 | 1.60 | 475 | 132 | 72 |
| B | 1 | 1.48 | 506 | 141 | 71 |

It can be seen that the use of 1/4 and 1/2 slot options in Scheme B gives slightly higher efficiency This would require the use of a multiplexed burst.

### *12.3.3.2 LCD384*

*Table 16: Modulation/coding schemes (LCD 384, Indoor A)*

| **Number of slots per frame** | **Modulation** | **C/I (dB)** | **Fraction of time used (re-use=3)** |
|---|---|---|---|
| 16 | B-OQAM | -1 (est) | 0.011 |
| 8 | B-OQAM | 2.0 | 0.028 |
| 6 | B-OQAM | 6.0 | 0.076 |
| 4 | Q-OQAM | 11.5 | 0.086 |
| 3 | Q-OQAM | 15.5 | 0.788 |

A 16 slot mode is needed to ensure coverage with re-use factor 3. The average number of slots per channel is 3.5, giving 12 channels per carrier. The Spectrum Efficiency is 5.8 Erlangs/Cell or 155kbps/MHz/Cell, with an average fractional loading of 49%. Clearly the small number of available channels limits the trunking efficiency.

## *12.4 Summary of Results*

| Environment | Service | Re-use factor | Spectral Efficiency (Erlangs/MHz/Cell) | Spectral Efficiency (kbps/MHz/Cell) |
|---|---|---|---|---|
| **Pedestrian A** | **Speech** | 1 | 38 | 152 |
| **Pedestrian A** | **LCD144** | 1 | 1.60 | 230 |
| **Pedestrian A** | **LCD144** | 3 | 1.74 | 250 |
| **Vehicular A** | **Speech** | 1 | 29 | 114 |
| **Vehicular A** | **Speech** | 3 | 57 | 226 |
| **Vehicular B** | **Speech** | 1 | 25 | 99 |
| **Vehicular A** | **LCD384** | 3 | 0.42 | 163 |
| **Indoor A** | **Speech** | 1 | 3.5 | 141 |
| **Indoor A** | **LCD384** | 3 | 0.40 | 155 |

The results shown above were obtained using the analytical method outlined earlier. The main simplifying assumptions are that the users are stationary, the system is noise limited, and that omni-directional antennas are deployed in hexagonal cells. This means that the modelling of the Pedestrian environment is compatible with the requirements of UMTS 30.03. The results in the other environments should be considered as indicative only.

# Annex D:
# Concept Group Delta $\delta$ - Wideband TDMA/CDMA

**This report contained in this annex was prepared during the evaluation work of SMG2 as a possible basis for the UTRA standard. It is published on the understanding that the full details of the contents have not necessarily been reviewed by, or agreed by, ETSI SMG or SMG2.**

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**Source: SMG2**

**TDoc SMG 897 / 97**

# Concept Group Delta WB-TDMA/CDMA: System Description Summary

## Introduction

In the procedure to define the UMTS Terrestrial Radio Access (UTRA), the WB-TDMA/CDMA concept group developed and evaluated a multiple access concept based on frequency, time, and code division.

The WB-TDMA/CDMA design rational is as follows:

- **CDMA component**: To offer interferer diversity, to provide fine granularity of user data rates without high peak to mean powers.
- **TDMA component based on GSM timing structure**: To build UTRA directly on top of proven GSM technology, to ensure easy handover between GSM and UMTS, to reduce the number of codes to be processed at the same time and hence make multi-user detection feasible from day 1 of UMTS. To take advantage of orthogonal partitioning of radio resources to avoid instability.
- **Benefit from near-far resistant multi user-detection (MUD)**: Cancellation of intra cell interference, to achieve stability without fast and accurate power control. To avoid soft handover.
- **Wideband carrier**: To support high user bit rates required in UMTS, and to take advantage of frequency diversity.

## Key technical characteristics of the basic system

**Table 1 - WB-TDMA/CDMA key technical characteristics**

| | |
|---|---|
| **Multiple Access Method** | TDMA and CDMA (FDMA inherent) |
| **Duplex Method** | FDD and TDD |
| **Channel Spacing** | 1.6 MHz |
| **Carrier chip / bit rate** | 2.167 Mchip/s |
| **Time slot structure** | 8 slots / TDMA frame |
| **Spreading** | Orthogonal, 16 chips/symbol |
| **Frame length** | 4.615ms |
| **Multi-rate concept** | multi-slot and multi-code |
| **FEC codes** | R = 1/8 . . . 1 (convolutional, punctured) |
| **Interleaving** | inter-slot interleaving |
| **Modulation** | QPSK / 16QAM |
| **Burst types** | 2 different burst types:<br>- burst 1: for long delay spread environments<br>- burst 2: for short delay spread environments |
| **Pulse shaping** | GMSK basic impulse $C_0(t)$ |
| **Detection** | coherent, based on midamble |
| **Power control** | slow |
| **Handover / IF handover** | mobile assisted hard handover |
| **Channel allocation** | slow and fast DCA supported |
| **Intracell interference** | suppressed by joint detection |
| **Intercell interference** | like in other clustered systems |

**Performance enhancing features (according to network operator choice)**

- BTS antenna hopping,
- frequency hopping,
- directive and/or adaptive antennas,
- time slot hopping,
- cell synchronization,
- interference based DCA,
- faster power control (for slow moving mobiles),
- quality based power control,
- co-channel multi-user detection (synchronized cells),
- relaying and advanced relay protocols such as ODMA.

As in GSM: In order to ensure that frequency hopping can be utilized in an operating system at any time, the frequency hopping capability shall be mandatory in all mobile terminals.

In addition, future enhancements of UTRA could include performance improving features which could take advantage of software configurable mobile station concepts, e.g. Turbo codes and improved link adaptation.

WB-TDMA/CDMA Support for Relaying and ODMA: A feasibility study conducted by the Delta and Epsilon concept groups concluded that WB-TDMA/CDMA can support relaying and the ODMA protocol with negligible increase in mobile complexity or cost. The ODMA protocol breaks difficult radio paths into a sequence of shorter hops which enables lower transmit powers or higher data rates to be used. It is the goal of the protocol to choose the least cost route through the relaying system when the relays are moving and the radio paths are dynamically changing. Simulations have shown that relaying has the potential to improve coverage and flexibility and may also increase capacity by lowering transmission powers and associated intercell interference. Further details are provided in part 6 of the WB-TDMA/CDMA Evaluation Report.

**System description**

Figure 1 depicts the multiple access of WB-TDMA/CDMA. For comparison reasons the multiple access of GSM is depicted in Figure 1, too. In WB-TDMA/CDMA up to 8 traffic channels (TCH) are supported in a time slot and a frequency band of width 1.6 MHz. However, the number of 8 traffic channels (CDMA codes) is not sharp. It is also feasible to assign 9 or even more CDMA codes within a time slot. The traffic channels are separated at the receiver based on TCH specific CDMA codes. As can be seen from Figure 1 in both WB-TDMA/CDMA and GSM a TDMA frame consists of 8 time slots. The duration of the TDMA frame as well as time slots is exactly the same for both GSM and WB-TDMA/CDMA. This identical frame and time slot structure allows to build WB-TDMA/CDMA easily on top of proven GSM technology and ensures an easy handover between GSM and UMTS.

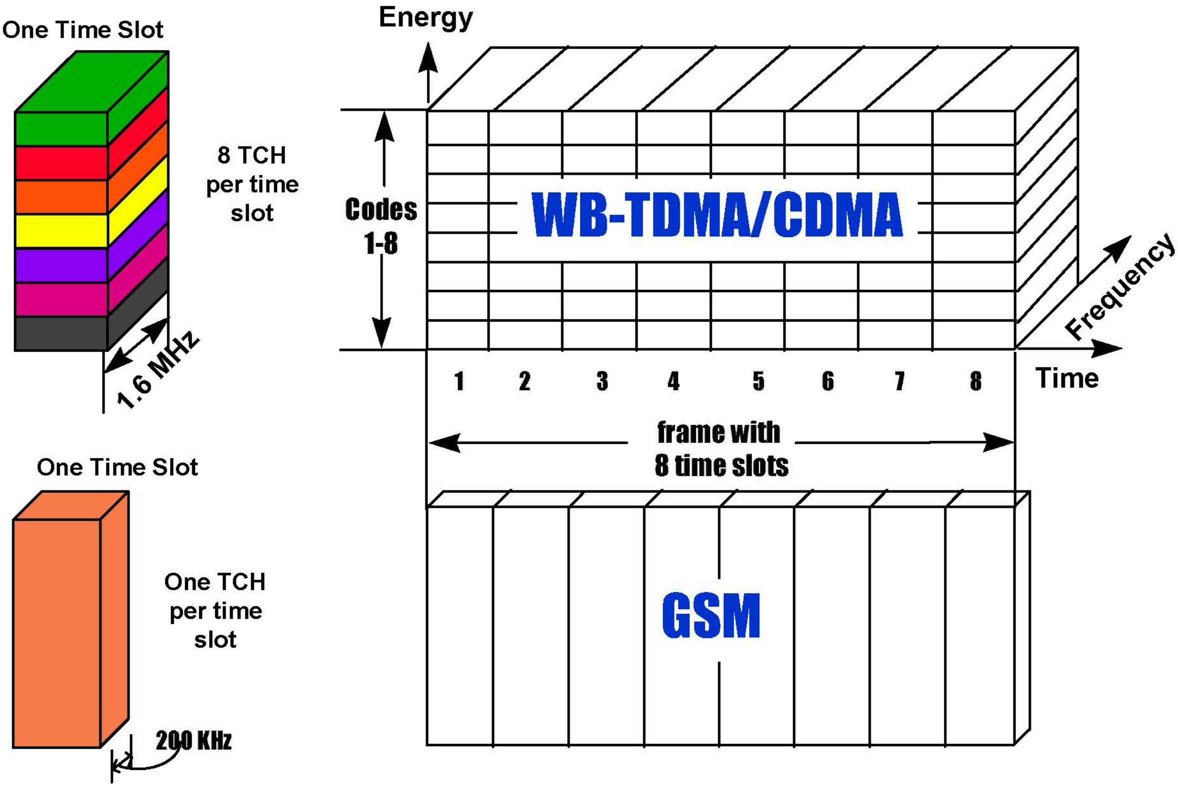


Figure 1: Multiple access of WB-TDMA/CDMA and GSM

In Figure 2 the uplink, i.e. the link from $K$ (e.g. equal to 8) mobiles to the base station, of WB-TDMA/CDMA is depicted. At the mobile $k$ ($k$ = 1,...,$K$) the data of the traffic channel $k$ is spread with the traffic channel specific CDMA code $k$. After spreading, the data of the traffic channel $k$ is put into 2 data blocks. As in GSM a WB-TDMA/CDMA transmitted burst consists of 2 data blocks and a midamble which is used for channel estimation. At the base station the received signal is the superposition of the bursts transmitted from all $K$ mobiles. At the base station receiver, first the different channels are estimated. This channel estimation uses as input the midambles assigned to the $K$ mobiles. Input for joint detection are the estimated $K$ radio channels as well as the $K$ CDMA codes assigned to the $K$ traffic channels. Then, in a single step multi-user / joint detection (JD) of data belonging to all $K$ traffic channels is performed. Intersymbol interference (ISI) and cross interference between data symbols of different traffic channels are eliminated. Due to the joint detection the requirements on uplink power control are quite relaxed and the benefits of CDMA without the intra-cell interference can be utilized. Furthermore, soft handover, which means additional infrastructure costs, is not needed.

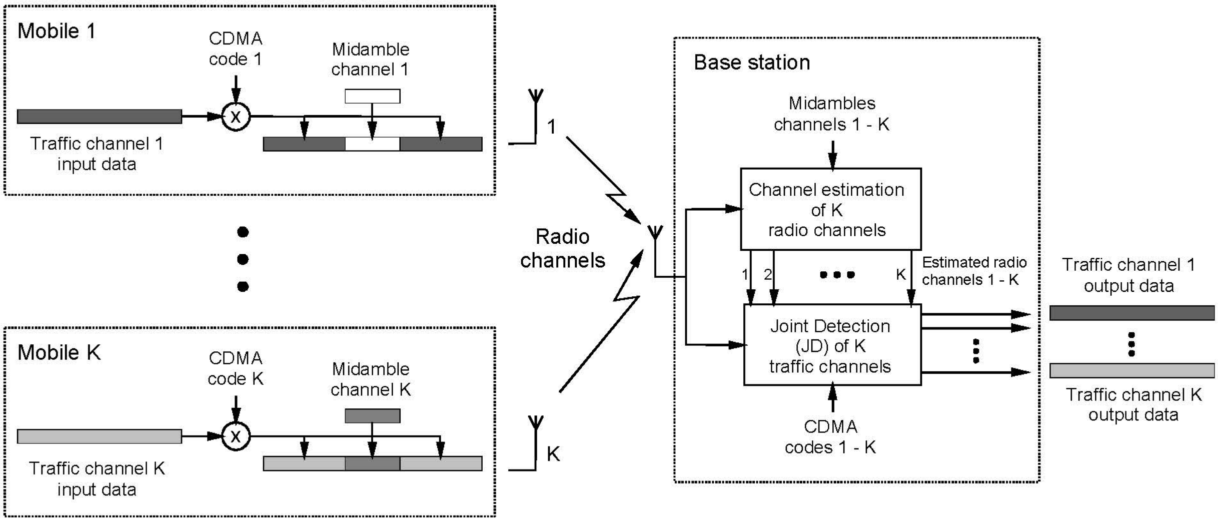


Figure 2: Uplink of WB-TDMA/CDMA

In the WB-TDMA/CDMA proposal user data rates from 8kbit/s up to 2Mbit/s with fine granularity can be adapted by

- assigning to a single user up to 8 time slots (multi-slot option),
- assigning to a single user up to 8 CDMA codes per time slot (multi-code option),
- adapting FEC coding rate (convolutional, punctured),
- adapting order of modulation (16QAM or QPSK).

For 16QAM data modulation the flexibility of data rates that can be provided to a single user is depicted in Figure 3. In Figure 3 the user data rate per CDMA code and per time slot is 32kbit/s.

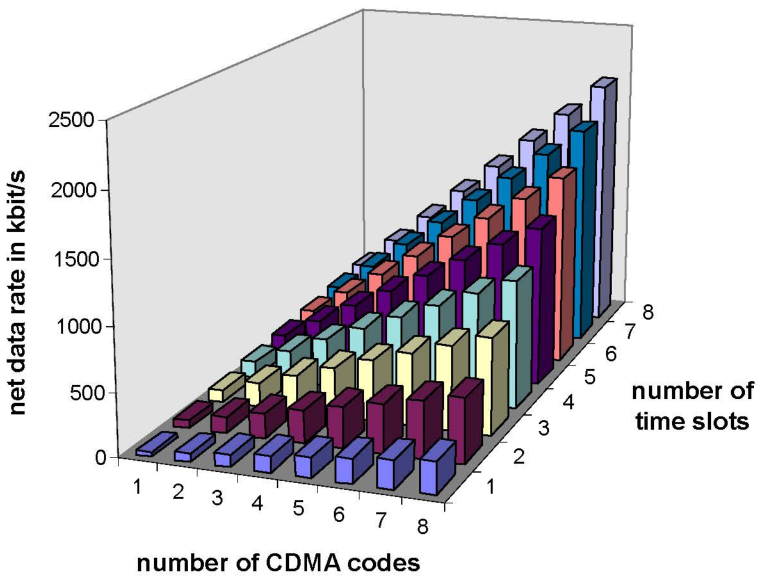


Figure 3: Flexibility of data rates in WB-TDMA/CDMA

**Summary**

The concept group delta WB-TDMA/CDMA proposal fulfills the UMTS requirements described in ETR 0401.

The main features can be summarized as follows:

- Support of hierarchical cell structures.
- Flexibility of user data rates by assigning different number of CDMA codes and time slots, adaptation of FEC code rate and used modulation type.
- Due to the moderate carrier bandwidth gradual introduction of UMTS islands on top of GSM networks is feasible.
- Stability of power control schemes guaranteed thanks to Multi User Detection.
- The WB-TDMA/CDMA proposal is an evolution towards wideband of the field proven GSM technology, i.e. keeping the same timing structure, and thus guarantees the robustness of the future UMTS system.
- Soft handover is not needed.
- Supports seamless handover between GSM and UMTS.
- The WB-TDMA/CDMA air interface used in unregulated frequency allocation for residential and business applications allows the use of uncoordinated systems through DCA techniques used in existing cordless systems.
- Some future enhancements are already foreseen to be introduced: adaptive antennas, new channel coding schemes, new receiver technologies, and ODMA.

**ETSI SMG#24**
**Madrid, Spain**
**December 15-19, 1997**
**Source: SMG2**

**TDoc SMG2 898 / 97**

# Concept Group Delta WB-TDMA/CDMA: Evaluation Summary

**Introduction**

In the procedure to define the UMTS Terrestrial Radio Access (UTRA), the WB-TDMA/CDMA concept group developed and evaluated a multiple access concept based on frequency, time, and code division.

The WB-TDMA/CDMA design rational is as follows:

- **CDMA component**: To offer interferer diversity, to provide fine granularity of user data rates without high peak to mean powers.
- **TDMA component based on GSM timing structure**: To build UTRA directly on top of proven GSM technology, to ensure easy handover between GSM and UMTS, to reduce the number of codes to be processed at the same time and hence make multi-user detection feasible from day 1 of UMTS. To take advantage of orthogonal partitioning of radio resources to avoid instability.
- **Benefit from near-far resistant multi user-detection (MUD)**: Cancellation of intra cell interference, to achieve stability without fast and accurate power control, to avoid soft handover.
- **Wideband carrier**: To support high user bit rates required in UMTS, and to take advantage of frequency diversity.

This document contains a brief summary how the concept group delta WB-TDMA/CDMA fulfills the high level requirements.

Please note, that part 6 of the evaluation report includes an addition to this summary showing how the ODMA enhancement can help to fulfill and exceed the high level requirements.

---

**maximum user bit rates**

- rural area: at least 144kbps (goal to achieve 384kbps), maximum speed is 500km/h
- suburban outdoor: at least 384kbps (goal to achieve 512kbps), maximum speed is 120km/h
- indoor/low range outdoor: at least 2Mbps, maximum speed is 10 km/h
- it is desirable that the definition of the UMTS air interface should allow evolution to higher bit rates

*Bit rates as requested in the high level requirements are well supported.*

*Real time 144kbps:*

- *allocating 1 code in each of the 8 time slots to a user (LCD 144a), QPSK,*
- *allocating 9 codes in 1 of the 8 time slots to a user (LCD 144b), QPSK,*
- *allocating 3 codes in 4 of the 8 time slots to a user (LCD 144c), QPSK.*

*Real time 384kbps:*

- *allocating 3 codes in each of the 8 time slots to a user (LCD 384a), QPSK,*
- *allocating 9 codes in 3 of the 8 time slots to a user (LCD 384b), QPSK.*

*Real time 2Mbps;*

- *9 codes are allocated in each of the 8 time slots to a user (LCD 2048), 16QAM.*

*Evolution to higher bit rates supported by e.g. higher RF bandwidth and/or higher order modulation.*

---

**flexibility**

- negotiation of bearer service attributes
- parallel bearer services (service mix), real-time/non-real-time communication modes
- adaptation of bearer service bit rate
- circuit and packet oriented bearers
- supports scheduling of bearers according to priority
- adaptation of link to quality, traffic and network load, and radio conditions
- wide range of bit rates should be supported with sufficient granularity

- variable bit rate real time capabilities should be provided
- bearer services appropriate for speech shall be provided

*High range of variability of user bit rates and bearer services due to*
- *pooling of time slots, pooling of CDMA codes,*
- *variation of modulation scheme,*
- *variation of FEC code rate, optimized combination of block and convolutional codes (outer and inner code)*

---

**handover**
- provide seamless handover between cells of one operator
- seamless handover between different operators or access network should not be prevented
- efficient handover between UMTS and 2nd generation systems, e.g. GSM, should be possible

*Seamless and efficient HO between both systems will be possible due to the same timing and frame structure in WB-TDMA/CDMA and GSM.*

*Soft handover is not used, thus HO between different operators or access networks has no performance loss.*

---

**compatibility with services provided by present core transport networks**
- ATM bearer services
- GSM services
- IP based services
- B/N-ISDN services

*Wide range of bearer classes will provide an efficient means of transport for core network services (ATM, GSM, IP, B/N-ISDN) over the radio interface.*

---

**radio access planning**
- if radio resource planning is required, automatic planning should be supported

*Radio resource planning for following items is necessary:*
- *Coverage, power, and frequency planning,*
- *planning of PICH (beacon frequency) and spreading codes.*

*Planning for the items listed above can be done in an automatic way.*

---

**public network operators**
- it shall be possible to guarantee pre-determined levels of quality-of-service and quality to public UMTS network operators, in the presence of other authorized UMTS users

*UMTS public operators (terrestrial as well as satellite) require dedicated frequency bands with appropriate guard bands.*

*The guaranteed pre-determined levels of QoS are met for:*
- *RT bearers with link adaptation (order of modulation, FEC code rate, optimized combination of block and convolutional codes (outer and inner code), number of physical channels used, etc.),*
- *NRT bearers with ARQ.*

*Network robustness is ensured by partitioning of radio resources.*

**private and residential operators**

- the radio access scheme should be suitable for low cost applications where range, mobility and user speed may be limited
- multiple non synchronized systems should be able to successfully coexist in the same environment
- it should be possible to install base stations without co-ordination
- frequency planning should not be needed

*It is recommended that private and public UMTS systems keep a separate frequency band.*

*Operating public and private systems in the same frequency band is possible by limiting the private systems' TX power and using DCA.*

*Operation in unpaired spectrum for unlicensed use possible due to inherent TDD capability.*

*TDD allows for simple low cost applications, where the radio resource is divided into independent units with fine granularity and thus, uncoordinated systems can coexist in the same geographical area using DCA.*

---

**spectrum efficiency**

- high spectrum efficiency for typical mixtures of different bearer services
- spectrum efficiency at least as good as GSM for low bit rate speech

*These requirements are very well supported by the WB-TDMA/CDMA proposal. For more details refer to part 4 of the evaluation report. Spectrum efficiency for speech is better than in GSM.*

---

**variable asymmetry of total band usage**

- variable division of radio resource between uplink and downlink resources from a common pool

*FDD:*

- *Overall traffic asymmetry requires larger downlink than uplink spectrum,*
- *single user traffic asymmetry is provided with assignment of different number of time slots and CDMA codes in uplink and downlink, respectively.*

*TDD:*

- *In the TDMA frame the switching point between uplink and downlink can be adapted dynamically,*
- *switching point dynamically set per cell,*
- *overall traffic asymmetry is supported in paired symmetric frequency bands.*

*Combination of FDD and TDD is possible.*

---

**spectrum utilization**

- allow multiple operators to use the band allocated to UMTS without coordination
- it should be possible to operate the UMTS in any suitable frequency band that becomes available such as first & second generation systems bands

*It is recommended that private and public UMTS systems keep a separate frequency band.*

*Operating public and private systems in the same frequency band is possible by limiting the private systems' TX power and using DCA.*

*Private and residential operators can use the same frequency band.*

*Operation in unpaired spectrum for unlicensed use possible due to inherent TDD capability.*

*The minimum required spectrum for re-farming is 3 x 1.6 MHz (reuse 3) + appropriate guard band.*

*Hot spot re-farming is possible with 1.6 MHz (single carrier) + appropriate guardband..*

*Relatively small carrier bandwidth which is an integer multiple of 200 kHz yields good re-farming granularity.*

---

**coverage, capacity**
- the system should be flexible to support a variety of initial coverage/capacity configurations and facilitate coverage/capacity evolution
- flexible use of various cell types and relations between cells within a geographical area without undue waste of radio resources
- ability to support cost effective coverage in rural areas

*Coverage / capacity evolution is possible due to adaptive antennas and DCA.*

*HCS is fully supported with at least 3 layers due to moderate bandwidth of the carriers.*

*Adaptation of frequency separated cell layers together with slow DCA are options to improve the capacity gains due to HCS.*

---

**mobile terminal viability**
- hand-portable and PCM-CIA card sized UMTS terminals should be viable in terms of size, weight, operating time, range, effective radiated power and cost

*RF linearity requirements are slightly higher as GSM today.*

*Required signal processing for joint detection is such that low cost terminals will be feasible day 1 when UMTS is introduced.*

---

**network complexity and cost**
- the development and equipment cost should be kept at a reasonable level, taking into account the cost of cell sites, the associated network connections, signaling load and traffic overhead

*Soft handover is not used, thus additional traffic and operating cost in the fixed network due to soft handover is avoided.*

*Smooth transition path for GSM networks is possible.*

*High TRX efficiency allows for small BTS.*

*If WB-TDMA/CDMA will be a world-wide standard, it is expected that the cost of base stations and associated equipment will benefit from a larger market. Interoperability between operators not only in Europe will also be much simpler since the core network will be based on GSM.*

---

**mobile station types**
- it should be possible to provide a variety of mobile station types of varying complexity, cost and capabilities in order to satisfy the needs of different types of users

*Low cost speech terminal:*
*1 time slot with only a 1 code capability in the uplink. and up to full code capability in the downlink for optional data applications.*

*Low cost terminal:*
*1 time slot with full code capability in the uplink and downlink for NRT services only.*

*Enhanced MS:*
*Duplex operation in every time slot and simultaneously monitoring of the surroundings*

*A duplexer is not needed below a certain level of time slot aggregation*

*Refer to the high level requirement mobile terminal variability, too.*

---

**alignment with IMT2000 (FPLMTS)**

- UMTS radio interface shall meet at least the technical requirements for submission as a candidate technology for IMT2000

*As WB-TDMA/CDMA meets the UTRA requirements, it can be submitted as IMT-2000 proposal.*

---

**minimum bandwidth allocation**

- it should be possible to deploy and operate a network in a limited bandwidth

*Uncoordinated operation of different UMTS operators within one frequency band requires appropriate guardbands between frequency allocations of each operator. The size of the necessary guardband is derived from the isolation between uncoordinated BTSs and MSs according to different scenarios considered in SMG2.*

*The minimum required spectrum for UMTS operators is 3 x 1.6 MHz (reuse 3) + guardband.*

---

**electromagnetic compatibility**

- the peak and average power and envelope variations have to be such that the degree of interference caused to other equipment is not higher than in today's systems

*The burst transmission due to the TDMA component is expected to cause similar EMC issues as in GSM.*

*In case of multi-code and / or 16QAM modulation additional envelope variations occur. However, it is expected that this can be tolerated, and EMC will not be degraded seriously.*

---

**RF radiation effects**

- UMTS radio interface shall be operative at RF emission power levels which are in line with the recommendations related to electromagnetic radiation

*In principal the average power levels of different mobile types are independent of SRTT.*

*The power levels of different mobile types can be specified such that recommendations are fulfilled.*

---

**security**

- UMTS radio interface should be able to accommodate at least the same level of protection as the GSM radio interface does

*This requirement is in principal independent from SRTT. Thus, the WB-TDMA/CDMA air interface can be specified such that level of protection is at least as that of the GSM radio interface.*

---

**coexistence with other systems**

- the UMTS radio interface should be capable to coexist with other systems within the same frequency band.

*Refer to explanations given for public, private and residential operators, as well as spectrum utilization.*

---

**dual mode**

- it should be possible to implement dual mode UMTS/GSM terminals cost effectively

*Harmonized approach between WB-TDMA/CDMA and GSM with respect to clocking, carrier spacing and frame structure.*

*For dual mode terminals additional GSM RX RF, and IF filters are required.*

*MLSE function in GSM can be realized by Joint Detection (JD) hardware.*

*No additional digital hardware for MLSE function in GSM is needed.*

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**Concept Group Delta**

**WB-TDMA/CDMA**

**System Description Performance Evaluation**

**Disclaimer:**

"This document was prepared during the evaluation work of SMG2 as a possible basis for the UTRA standard. It is provided to SMG on the understanding that the full details of the contents have not necessarily been reviewed by, or agreed by, SMG2."

**ETSI SMG2#24** **TDoc SMG2 368 / 97**

**Cork, Ireland**

**December 1-5, 1997**

**Concept Group Delta**

**Wideband TDMA/CDMA**

**Evaluation Report - Part 1**

**V 2.0 b**

# Table of Contents

# 1 Introduction

In the procedure to define the UMTS Terrestrial Radio Access (UTRA), the wideband TDMA/CDMA concept group develops and evaluates a multiple access concept based on time, frequency and code division. This group basically includes the FRAMES Multiple Access Mode 1 (FMA1) with spreading proposal.

Intention of this document is to present detailed technical data and evaluation results for the concept discussed in this group.

The final version 2.0 b of the evaluation report contains 6 parts:

Part 1: Description of the proposal

Part 2: Mixed services in Wideband TDMA/CDMA

Part 3: Link level simulation results

Part 4: System level simulation results

Part 5: Template according UMTS 30.03, annex 1

Part 6: Support for Relaying and ODMA

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## 2 Abbreviations

| | |
|---|---|
| 16QAM | 16-ary Quadrature Amplitude Modulation |
| ARQ | Automatic Repeat on Request |
| BCCH | Broadcast Control Channel |
| BER | Bit Error Rate |
| BPSK | Binary Phase Shift Keying |
| BS | Base Station |
| BSS | Base Station Subsystem |
| CA | Capacity Allocation |
| CAA | Capacity Allocation Acknowledgement |
| CBR | Constant Bit Rate |
| CCCH | Common Control Channel |
| CD | Capacity Deallocation |
| CDA | Capacity Deallocation Acknowledgement |
| CDMA | Code Division Multiple Access |
| CTDMA | Code Time Division Multiple Access |
| CRC | Cyclic Redundancy Check |
| CU | Central Unit |
| CUCH | Common Uplink Channel |
| DCA | Dynamic Channel Allocation |
| DCCH | Dedicated Control Channel |
| DL | Downlink |
| DS | Direct Sequence |
| DTX | Discontinuous Transmission |
| DU | Data Units |
| FACCH | Forward Associated Control Channel |
| FACH | Forward Access Channel |
| FCCH | Frequency Correction Channel |
| FCH | Frame Control Header |
| FDD | Frequency Division Duplex |
| FDMA | Frequency Division Multiple Access |
| FEC | Forward Error Control |
| FER | Frame Error Rate |
| FMA1 | FRAMES Multiple Access Mode 1 |
| FO | Forward Order |
| FOCH | Forward Order Channel |
| FRAMES | Future Radio wideband Multiple Access |
| FWA | Fixed Wireless Access |
| GMSK | Gaussian Minimum Shift Keying |
| HCS | Hierarchical Cell Structure |
| IA | Interference Averaged |
| JD | Joint Detection |
| L1 | Layer 1 |
| L2 | Layer 2 |
| LLC | Logical Link Control |
| MA | Multiple Access |

| | |
|---|---|
| MAC | Medium Access Control |
| MAHO | Mobile Assisted Handover |
| MOHO | Mobile Originated Handover |
| MS | Mobile Station |
| | |
| NRT | Non-Real Time |
| | |
| PC | Power Control |
| PCH | Paging Channel |
| PICH | Pilot Channel |
| POTS | Plain Old Telephone Service |
| PWCCH | Public Power Control Channel |
| | |
| QAM | Quadrature Amplitude Modulation |
| QOQAM | Quaternary Offset Quadrature Amplitude Modulation |
| QoS | Quality of Service |
| QPSK | Quaternary Phase Shift Keying |
| | |
| RACH | Random Access Channel |
| RAU | Remote Antenna Units |
| RF | Radio Frequency |
| RLC | Radio Link Control |
| RRC | Radio Resource Control |
| RRM | Radio Resource Management |
| RT | Real Time |
| | |
| SCH | Synchronization Channel |
| SDCCH | Stand-alone Dedicated Control Channel |
| | |
| TCH | Traffic channel |
| TDD | Time Division Duplex |
| TDMA | Time Division Multiple Access |
| | |
| UACH | Uplink Acknowledgement Channel |
| UL | Uplink |
| UMTS | Universal Mobile Telecommunications System |
| | |
| VBR | Variable Bit Rate |

# 3 Logical channels

This chapter describes the logical channels required for data transfer. Logical channels are unidirectional and can be divided into two categories:

- Traffic channels and
- Control channels

## 3.1 Traffic channels

A traffic channel (TCH) is used for transferring user data and / or layer 3 signalling data.

## 3.2 Control channels

Control channels carry layer 3 and MAC signalling data and they are also needed for the initial synchronization of the mobile station. Control channels can be further divided into:

- Dedicated control channels and
- Common control channels.

### 3.2.1 Dedicated Control Channels

Dedicated Control Channels (DCCH) are point-to-point control channels that carry connection oriented messages.

1. Associated Control Channel (ACCH) is a point-to-point channel in the uplink or downlink direction. The allocation of an ACCH is linked to the allocation of a TCH. The ACCH is used for RLC/MAC layer messages, e.g. capacity allocations or link control messages.
2. Stand-alone DCCH (SDCCH) is a point-to-point channel in the uplink or downlink direction. The allocation of an SDCCH is not linked to the allocation of a TCH, MAC may or may not allocate SDCCH capacity to an MS dependent upon circumstances. An SDCCH is used for the transfer of layer 3 and RLC/MAC layer messages.

### 3.2.2 Common Control Channels

Common Control Channels (CCCH) are point-to-multipoint or point-to-point control channels that carry connectionless or connection oriented messages.

PICH Pilot Channel is a point-to-multipoint channel in the downlink direction. The PICH is used for power measurements and initial frequency and time slot synchronization by the mobile stations.

BCCH Broadcast Control Channel is a point-to-multipoint control channel in the downlink direction. The BCCH is used for the broadcast of layer 3 and MAC information that describes the cell.

PCH Paging Channel is a point-to-multipoint control channel in the downlink direction. The PCH is used for the broadcast of layer 3 paging messages.

RACH Random Access Channel is a contention access uplink channel that can be used by MS to signal a number of messages, e.g. capacity request messages and access request messages. The RACH may be segmented into two components, one for use by MS that have time alignment with the cell (this component makes use of a normal burst (not access burst) and is therefore called N-RACH) and one for use by MS that do not have time alignment with the cell (this component uses a short access burst and is called S-RACH. There exists two types of such access bursts, depending on the cell size).

FACH Forward Access Channel is a point-to-multipoint channel in the downlink direction that is used to transfer MAC related signalling (e.g. capacity allocations).

UACH Uplink Assigned Channel is a point-to-point uplink channel that is temporarily assigned to an MS either for the acknowledging of certain MAC messages or for data unit transmission requests for downlink NRT data transfer.

SCH Synchronization channel is a point-to-multipoint downlink channel that is used by the mobile station to synchronize with the TDMA multiframe structure of the BS.

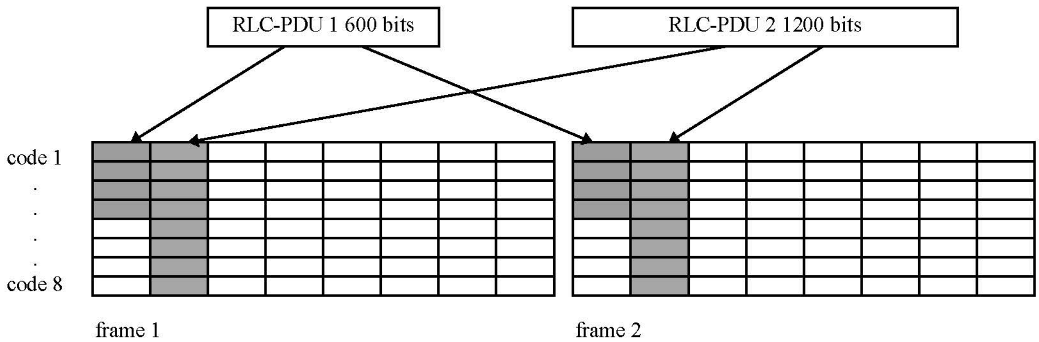
## *3.3 Mapping of Logical Channels to Physical Channels*

This section describes the way in which logical channels are mapped onto physical resources. Details on the multiframe structure are given in section 4.1.

In the sequel, we use the terms physical channel and resource unit; a physical channel is defined as the association of one time slot and one frequency (or set of frequencies + hopping sequence when slow frequency hopping is used). A resource unit is that part of a physical channel that is associated with one spreading code. A physical channel therefore comprises up to m resource units where m is the maximum number of available codes in one time slot.

### 3.3.1 Traffic Channels

A traffic channel is allocated one or more sets of slots/codes within a frame together with an interleaving interval. Each set of slots and codes over an interleaving interval maps to a data unit and a data unit can correspond to an FEC code block and RLC protocol data unit. This is illustrated by the following diagram:




For RT allocation, a traffic channel is mapped onto one or more resource units over an indeterminate period of time. A release procedure is necessary to liberate the resource. The mapping of a TCH on slots and codes can be each TDMA frame, every second TDMA frame, every fourth TDMA frame,...up to every 64th TDMA frame.

For NRT allocation, the mapping of the traffic channel onto resource units is valid only for a so-called allocation period.

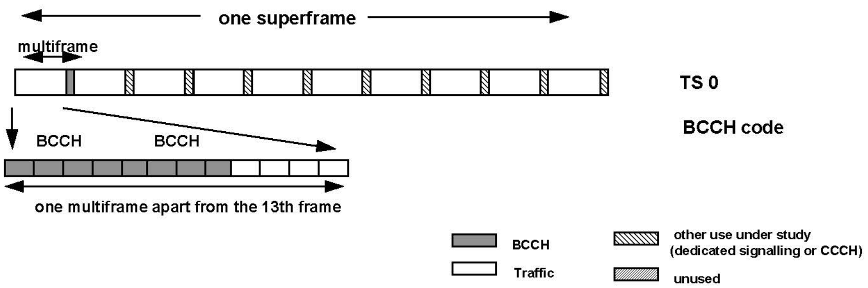
### 3.3.2 Control Channels

**The Pilot Channel PICH** is transmitted at a constant power by the BS on the BCCH carrier. In FDD mode, it is transmitted continuously (on all time slots per frame). In TDD mode, it is transmitted only during the downlink time slots. The PICH is used by the mobile stations to perform power measurements (for cell (re-)selection and mobile assisted handover) and to acquire initial frequency and time slot synchronization (time slot boundary).

The Pilot Channel is based on a long chip-sequence hidden under the data channels by large spreading; the transmitted power of the PICH is expected to be 16 dB below the highest allowed power of any downlink data channel. The long chip sequence used for synchronization matches the time-slot length of 1250 chips. Eight orthogonal sequences are defined. They are allocated on a base station basis to control the risk of interference between close cells sharing the same frequency.

At power up or during the monitoring of adjacent cells, the mobile station scans for one of the eight possible sequences over windows of 2 x 1250 chips to determine the time and frequency synchronization. The result of the autocorrelation is a good estimate of the received power level. It must be pointed out that the eight sequences can be simultaneously analysed, which may allow the mobile station, from the same set of samples, to get power measurements and synchronization from several base stations sharing the same BCCH frequency.

**The Broadcast Control Channel BCCH** is mapped onto a predefined time-slot (e.g. time-slot 0) and predefined spreading code, hereafter called BCCH spreading code. The BCCH logical channel may occupy partly or fully the corresponding resource unit, apart from the 13th frames carrying SCH, dedicated signaling or idle frames, as explained later. This is illustrated in the figure below. Additional resource units located anywhere in terms of slot or frequency may carry additional BCCHs. The exact position of these additional channels is indicated in the “primary” BCCH (e.g. on TS 0).




**Mapping of the BCCH logical channel**

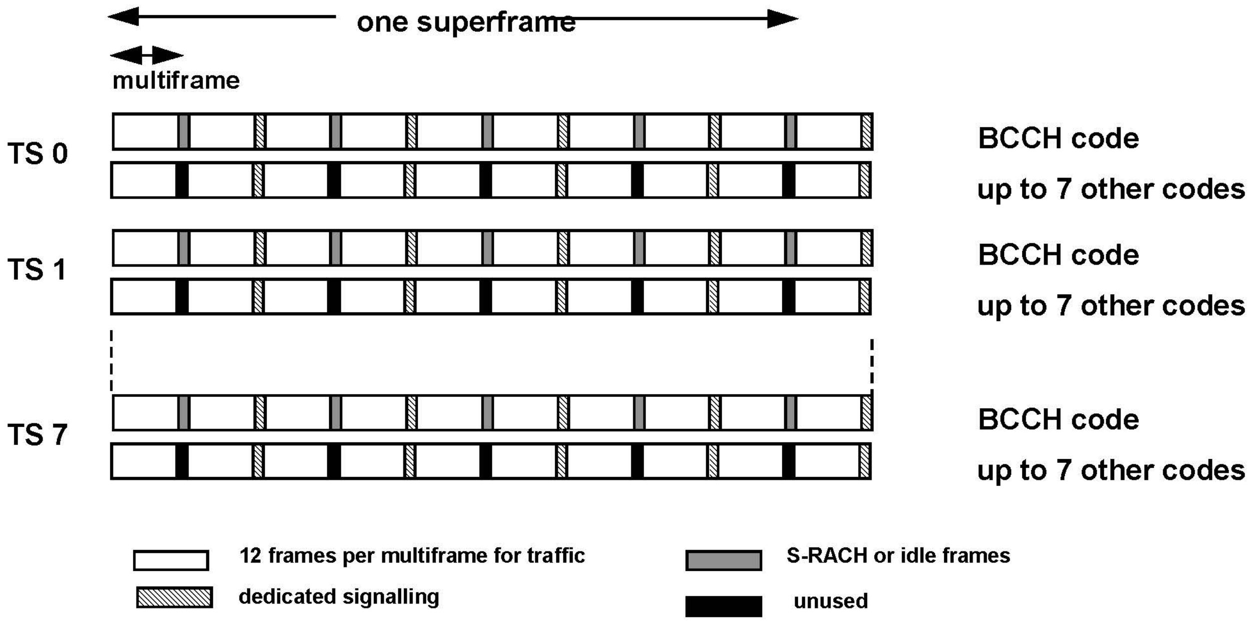
**The Paging Channel PCH** can be mapped on any combination of time slots and codes on the BCCH carrier, and it can occupy partly or fully the correspondingresource units, apart from the 13th frames carrying SCH or dedicated signalling. In case other frequencies carry additional BCCH, they can also carry additional PCH. The exact location of the PCH is indicated on the BCCH. Of course, the chosen location must allow efficient DRX. One possibility is to map PCH on TS 0 of the BCCH carrier and on the BCCH spreading code (BCCH and PCH thus sharing the sameresource unit).

**The Random Access Channel RACH** (both N-RACH and S-RACH) has an interleaving period of one frame and each transmission occupies only one burst.

The N-RACH is an uplink contention access signalling channel for MS being time-aligned that is used to transfer MAC related signalling to the network (e.g. capacity requests). It can be mapped on anyresource unit and it can occupy it partly or fully, apart from the 13th frames. Its location can be indicated on the BCCH or on the FACH. The total N-RACH capacity may be subdivided amongst a number of access groups.

The S-RACH is used by a mobile station that needs to access the network without knowing its timing advance (either on initial access or for handover access in the case of asynchronous handovers). Characteristics of the joint detection impose that S-RACH bursts are not transmitted simultaneously to traffic bursts. A fixed (basic) mapping on the multiframe structure can be used, whereby the S-RACH are mapped on some of the 13th frames on each time slot and on each carrier; as an example, the S-RACH can be mapped on 5 multiframes out of the 9 multiframes of a superframe, as shown in the next figure. This mapping applies both to FDD and TDD modes, but, in TDD, only the uplink time slots can be used.

Of course, other dynamic or extended mappings are possible; for example, in order to increase the S-RACH capacity, time slots of the 12 first frames of the multiframe may be partly or fully allocated in the uplink for random access opportunities.




**Mapping of the basic S-RACH on all time-slots and all carriers for the fixed mapping case**

Since transmitting random accesses and monitoring neighbouring cells can be considered as two mutually exclusive actions, the 13th uplink frames reserved for S-RACH in the fixed mapping case are also the "idle" frames. The corresponding downlink frames are also idle.

**The Forward Access Channel FACH** can be mapped on any combination/fraction of downlinkresource units. Again, the 13th frames cannot be used for the FACH. The FACH can be segmented into a number of parallel channels each serving a group of MS if required.

**The Uplink Assigned Channel UACH** can be allocated anyresource unit on the uplink (not on the 13th frames), but the allocation is made for the transfer of only one message (e.g. acknowledgement or forward order message).

**The Synchronization Channel SCH** uses the synchronization burst format. The exact burst format is still under study. However in principle it must fulfil the same role as the synchronization burst in GSM. This burst carries the base station identification and the frame number, in order to locate the frame within the multiframe and superframe. The channel coding is such that a single burst carries the whole information. There is no interleaving between bursts.

The SCH channel is mapped as a minimum onto the BCCH carrier but may also be mapped on any carrier, not only the carrier on which the pilot channel is transmitted.

If Y and W are non zero positive integers, the SCH channel can be found Y times per superframe (where the superframe comprises Y+4W multiframes as described in section 4.1) on any of the 8 time-slots (TS=0 to 7). The SCH occupies part of the "signalling" frames positions, i.e the 13th frames positions.

An example of superframe comprising 9 multiframes is obtained with Y = 1 and W = 2; in that case, the SCH appears once per superframe for any time slot of the BCCH carrier. Hence for the time-slot number TS, the SCH should appear at the frame with number FN={ (TS+1)*13} mod (13*9). In effect this means that an SCH slot appears every 13 frames in one time-slot, apart from the end of the superframe that exhibits a gap of 26 frames without SCH. Most of the time, an SCH appears roughly every 60ms. This scheme applies both to FDD and TDD modes, but in TDD mode, only the downlink time slots can be used.

When the SCH is transmitted, nothing else is transmitted in the same time slot on the remaining available spreading codes. The absence of transmission of traffic burst on the BCCH carrier on frames with synchronization bursts is due to some characteristics of the joint detection. The mapping of the SCH is illustrated in Figure 3-1.

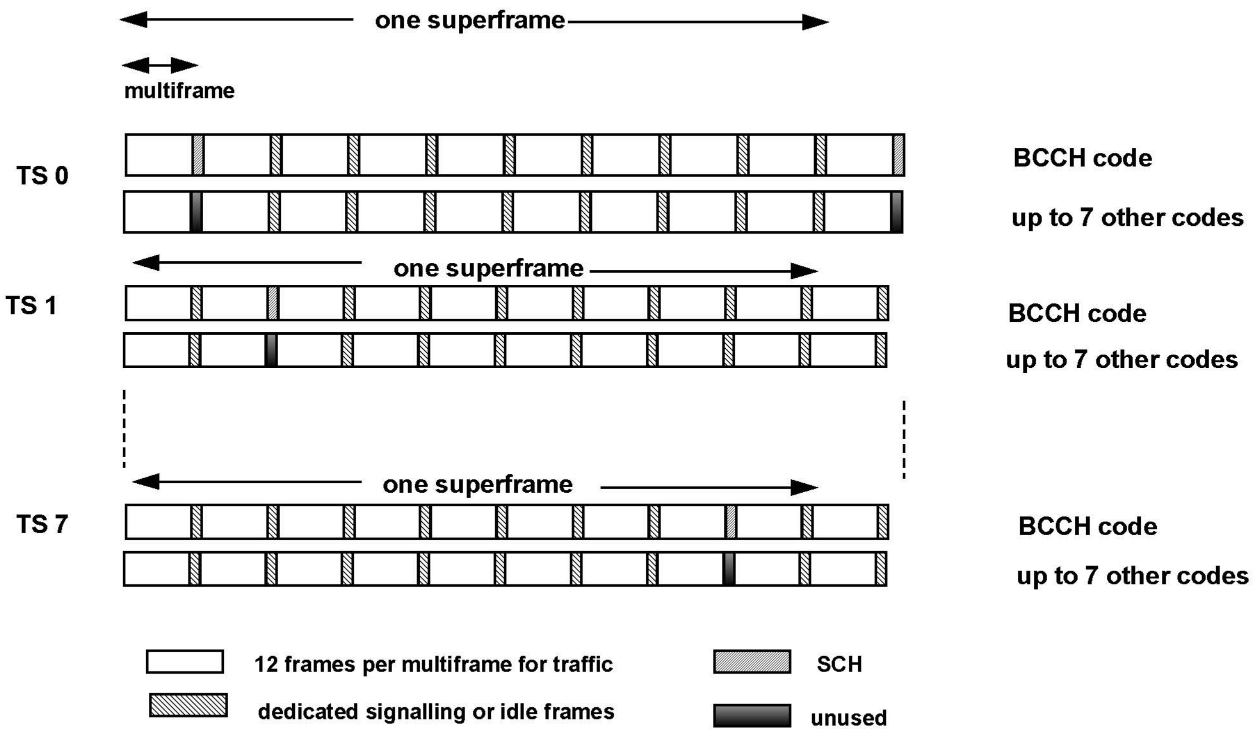



*Figure 3-1: Mapping of the SCH channel*

**The Associated Control Channel ACCH** is mapped, either on the 13th frames that are not idle or used for SCH or S-RACH signalling, or by stealing capacity allocated to a TCH for a bearer service. In case of stealing, the ACCH can steal capacity from any TCH allocated for the same MAC identifier to which the MAC message is addressed.

**The Stand-alone Dedicated Control Channel SDCCH** can be mapped on anyresource unit and can occupy it partly or fully, apart from the 13th frames. The allocation is usually made for a relatively short period of time.

# 4 Physical channel structure

TD/CDMA can operate in FDD mode and in TDD mode. The channel spacing of TD/CDMA is 1.6 MHz both in FDD and in TDD mode. The basic physical channel of TD/CDMA is a certain time slot and one spreading code on a certain carrier frequency (code physical channel). In the following, an overview about the multiframe, unit frame, time slot and code structure is given. Then, the modulation method is defined. Finally, an example of service mappings to physical channels is given.

## *4.1 Multiframe*

The requirements for the multiframe structure come from two major directions. First, it is required that seamless handovers between TD/CDMA and GSM can be made. This implies a GSM like multiframe structure that can be further improved by taking into account the identified deficiencies in GSM. Secondly, the control requirements from the packet access protocol (RLC/MAC) have to be incorporated into the multiframe structure. One candidate multiframe structure is proposed next and shown in *Figure 4-1*.

In the proposed structure, allresource units (all carriers, all time-slots, all the codes per time-slot) use a 13-frame multiframe. If aresource unit is allocated to a traffic channel, 12 out of 13 frames (both in up and downlink) carry the user information, the remaining frame being used either for dedicated (ACCH), common control signalling (SCH or S-RACH) or monitoring. This is illustrated in Figure 4-1 where the "signalling" frame corresponds in a static way to the 13th frame. Dynamic mapping of the "signalling" frame may be investigated later.

Also theresource unit(s) carrying the BCCH carrier use that 13 frame multiframe. This is analogous to the GSM 26 frame multiframe, where half of the 13th frame are used for the SACCH and half for the monitoring (idle frame).

The mapping of logical channels onto that multiframe structure is given in section 3.3.

A superframe comprises several, say X, multiframes. As an example we chose a superframe made of X=9 multiframes, hence 117 frames, as illustrated in Figure 4-1. In general X should be equal to Y+4W, where Y and W are non zero positive integers; the example of Figure 4-1 corresponds to Y = 1 and W = 2.

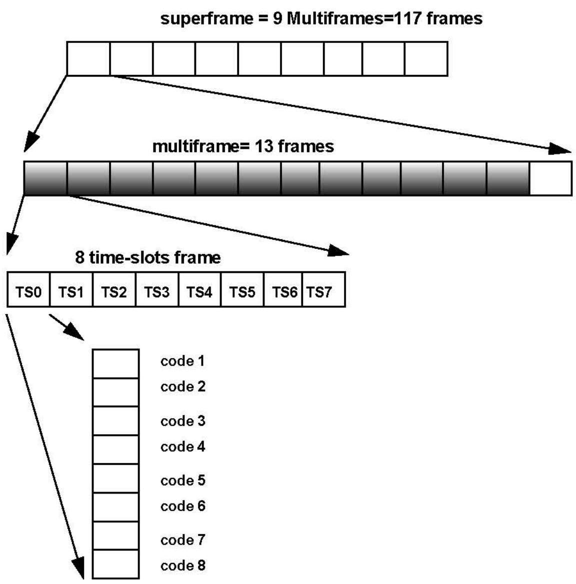



*Figure 4-1 A candidate multiframe structure that aims to provide compatibility with the GSM multiframe structure in order to make handovers between TD/CDMA and GSM possible.*

### 4.2 Frame Structure

In the following sections, a unit frame structure is outlined.

#### 4.2.1 Time slots

The TDMA frame is subdivided into eight time slots of 577 µs duration each. This interval corresponds to 1250 chip periods. The physical content of the time slots are the bursts of corresponding length as described in Section 4.3.

#### 4.2.2 FDD frame

The unit FDD frame is presented in Figure 4-2. The length of the FDD frame is 4.615 ms which is 10000 chip periods.

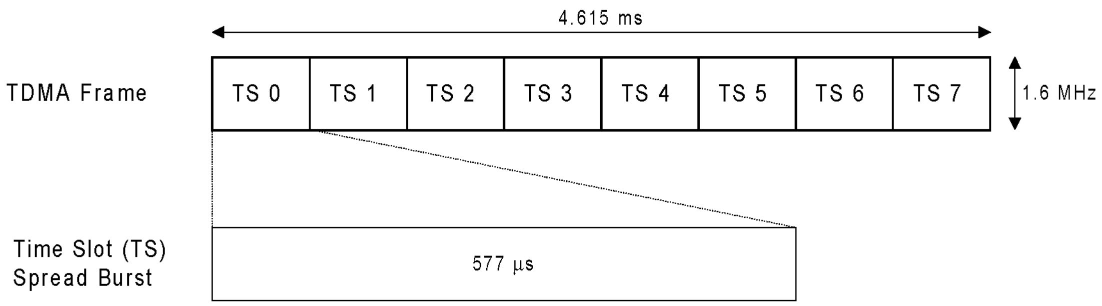



*Figure 4-2 The unit FDD frame structure of WB-TDMA/CDMA.*

#### 4.2.3 TDD frame

The TDD frame is of the same length as the FDD frame but it is divided into downlink and uplink parts (Figure 4-3). The switching point between uplink and downlink can be moved in the TDD frame to adopt asymmetric traffic. The minimum length of uplink and downlink parts is one eighth of the frame length (577 µs).

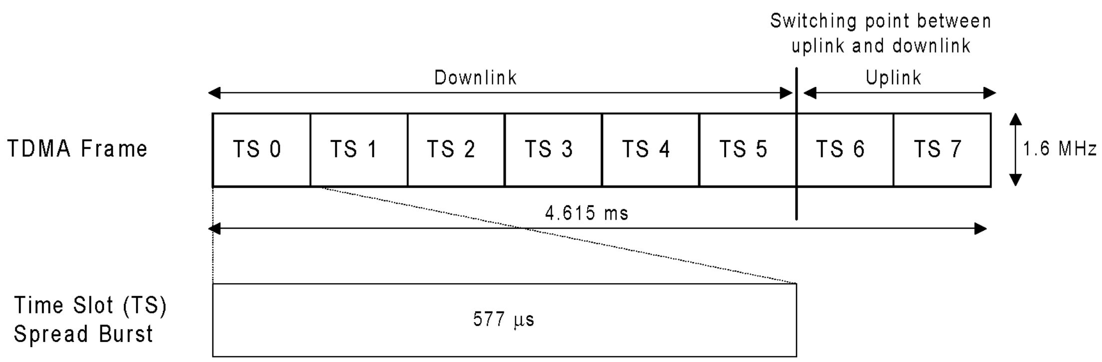



*Figure 4-3 The unit TDD frame structure of WB-TDMA/CDMA.*

In the TDD frame structure, it is assumed that the same mobile station is not receiving in the last slot of the downlink part and transmitting in the first slot of the uplink part.

#### 4.2.4 Spreading codes

Within each time slot of length 577 µs, an additional separation of user signals by spreading codes is used. This means that within one time slot of length 577 µs, more than one burst of corresponding length as described in Section 4.3 can be transmitted. These multiple bursts within the same time slot can be allocated to different users or partly or all to one and the same user. For the multiple bursts within the same time slot, different spreading codes are used to allow the distinction of the multiple bursts.

The spread bursts as described in Section 4.3 are designed in such a way that up to 8 bursts can be transmitted within one time slot in case the bursts are allocated to different users in the uplink. In the downlink as well as in case all bursts within one time slot are allocated to one and the same user in the uplink, also more than 8 bursts (e.g. 9 or 10) can be transmitted within one time slot. By the transmission of multiple bursts within one time slot, small bit rate granularity and high bit rate flexibility are achieved, thus allowing the implementation of the whole service range from low to high bit rates.

### *4.3 Bursts*

#### 4.3.1 Traffic bursts

Two types of traffic bursts are defined: the Spread Speech/Data burst 1 (S1) and the Spread Speech/Data burst 2 (S2). The Speech/Data bursts 1 and 2 consist of two data symbol fields, training sequence field and guard period (Figure 4-4 and Figure 4-5). The training sequence length of the Spread Speech/Data burst 1 is 296 chip periods long whereas the training sequence length of the Spread Speech/Data burst 2 is 107 chip periods long. A set of training sequences is defined in Section 4.4.3.

Both burst formats can be used for all services from speech to high rate data up to 2 Mbit/s. The midamble length of the Spread Speech/Data burst 1 is suited for estimating the different uplink channel impulse responses of 8 users within the same time slot with a time dispersion of up to about 15 µs. If the number of users is reduced, the tolerable time dispersion is increased. For instance, the midamble length of the Spread Speech /Data burst 1 is also suited for estimating the different uplink channel impulse responses of 4 users within the same time slot with a time dispersion of up to about 25.5 µs. The midamble length of the Spread Speech/Data burst 1 is also suited for estimating the downlink channel impulse response with a time dispersion of 53.5 µs, independent of the number of active users; furthermore, for estimating the uplink channel impulse response with a time dispersion of up to about 53.5 µs in case all bursts within a slot are allocated to one and the same user. However, in this case some symbols of the data symbol field following the midamble might have to be punctured in order to increase the guard interval. The midamble length of the Spread Speech/Data burst 2 is suited for estimating the different uplink channel impulse responses of 8 users within the same time slot with a time dispersion of about 5.5 µs; furthermore, for estimating the downlink channel impulse response with a time dispersion of up to about 15 µs and higher, independent of the number of active users; furthermore, for estimating the uplink channel impulse response with a time dispersion of up to about 15 µs and higher in case all bursts within a slot are allocated to one and the same user.

Thus, the Spread Speech/Data burst 1 can be used for

- uplink, up to 8 different users per time slot, channel time dispersion of up to about 15 µs, all services from speech up to 2 Mbit/s,
- uplink, in case the bursts within a time slot are allocated to up to 4 different users, channel time dispersion of up to about 25.5 µs, all services from speech up to 2 Mbit/s, (example)
- downlink, independent of the number of active users, channel time dispersion of up to 53.5 µs, all services from speech up to 2 Mbit/s,
- uplink, in case all bursts within a slot are allocated to one and the same user, channel time dispersion of up to 53.5 µs, all services from speech up to 2 Mbit/s.

The Spread Speech/Data burst 2 can be used for

- uplink, up to 8 different users per time slot, channel time dispersion of up to about 5.5 µs, all services from speech up to 2 Mbit/s,
- downlink, independent of the number of active users, channel time dispersion of up to 15 µs and higher, all services from speech up to 2 Mbit/s,
- uplink, in case all bursts within a slot are allocated to one and the same user, channel time dispersion of up to 15 µs and higher, all services from speech up to 2 Mbit/s.

Concerning the use of the different bursts, confer also Section 4.4.3.

The payloads (number of data symbols) of the Spread Speech/Data bursts 1 and 2 are 56 symbols and 68 symbols, respectively. The use of the individual symbols is defined in Table 4-1and Table 4-2.

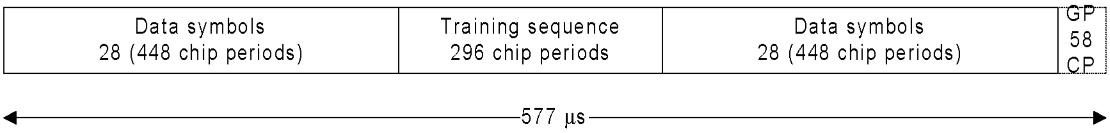



*Figure 4-4 Burst structure of the Spread Speech/Data burst 1. GP stands for guard period and CP for chip periods.*

*Table 4-1 The contents of the Spread Speech/Data burst 1 fields and the use of individual chips.*

| Chip number (CN) | Length of field in chips | Length of field in symbols | Contents of field |
|---|---|---|---|
| 0-447 | 448 | 28 | Data symbols |
| 448-743 | 296 | - | Training sequence |
| 744-1191 | 448 | 28 | Data symbols |
| 1192-1249 | 58 | - | Guard period |

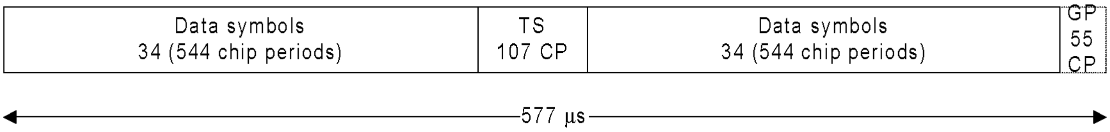



*Figure 4-5 Burst structure of the Spread Speech/Data burst 2. TS stands for training sequence, GP for guard period and CP for chip periods.*

*Table 4-2 The contents of the Spread Speech/Data burst 2 fields and the use of individual chips.*

| Chip number (CN) | Length of field in chips | Length of field in symbols | Contents of field |
|---|---|---|---|
| 0-543 | 544 | 34 | Data symbols |
| 544-650 | 107 | - | Training sequence |
| 651-1194 | 544 | 34 | Data symbols |
| 1195-1249 | 55 | - | Guard period |

### 4.3.2 Access Bursts

Two proposals of RACH bursts are presented below, allowing either to maximise the admissible delay spread by putting a long guard period at the end of the burst, either to maximise the capacity of the access channel in low delay spread environments by multiplying by two the number of admissible RACH in a timeslot.

Two bursts are defined, a short one with a length of 625 chips (288.5 ms) and a long one of 1250 chips (577 ms). Their structure is precised by the two figures below :

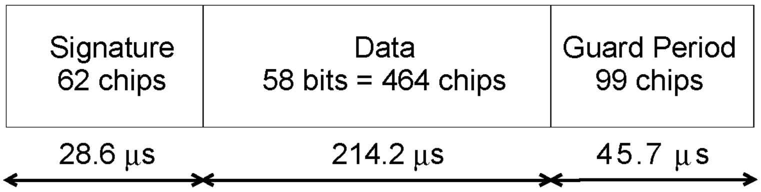



Figure 1 : Short access burst

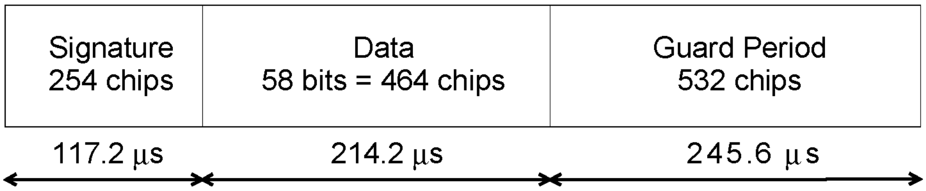



Figure 2 : Long access burst

The long burst exactly matches the timeslot length. The guard period is sufficiently long to allow for a cell radius of 35 km offering in addition 11.7 us of tolerance for delay spread and imperfect mobile synchronisation.

The short burst is a half of the timeslot length so that a timeslot allocated for access can be subdivided in two, this scheme doubles the number of access channels in areas with a low-delay spread, typically the indoor environment.

The base station indicates in the BCCH the expected kind of access burst.

These bursts are defined so that the random access channel offers the highest possible throughput by allowing to resolve whenever possible the concurrent accesses and otherwise to pick up the best signals while ignoring the others. For this reason the access burst carries enough information :

- to statistically allow a discrimination between several mobiles simultaneously attempting an access, consequently allowing to unambiguously inform each of them when its message was correctly decoded and in this case which was the channel allocated;
- for the mobile to give indication of the requested service, to allow a direct channel allocation without extra exchanges on a dedicated physical channel that would result in a delayed call establishment.

### 4.3.3 Pilot burst

The pilot burst is made of 1250 chips in order to exactly match the timeslot duration.

## *4.4 Modulation*

In this chapter, there has been made a separation between the data modulation and the spreading modulation. The data modulation is defined in Section 4.4.1 and the spreading modulation in Section 4.4.2. The basic modulation parameters including pulse shaping are summarized in Table 4-3.

*Table 4-3 Basic modulation parameters*

| | |
|---|---|
| Carrier symbol/chip rate | 2.167 Mchip/s |
| Carrier spacing | 1.6 MHz |
| Data modulation | QPSK<br>16QAM |
| Spreading modulation | Linearised GMSK |
| Spreading characteristics | Orthogonal<br>16 chips/symbol |

### 4.4.1 Data modulation

In this section, symbol rates and durations are defined and the mapping of bits onto signal point constellation is shown.

#### *4.4.1.1 Symbol rate*

The symbol rates and symbol durations are summarized in Table 4-4.

*Table 4-4 Summary of WB symbol rates and durations.*

| Symbol rate | Symbol duration |
|---|---|
| 135.41 ksymbol/s | 7.384 µs |

#### *4.4.1.2 Mapping of bits onto signal point constellation*

In WB-TDMA/CDMA 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 in the same frequency band of width B equal to 1.6 MHz. The maximum possible number of CDMA codes, which is smaller or equal to 16, depends on the actual interference situation and the service requirements. In Section 4.3.1 the bodies of such spread bursts associated with a particular user are shown. Each user burst has 2 data carrying parts termed data blocks

$$\underline{\mathbf{d}}^{(k,i)} = (\underline{d}_1^{(k,i)}, \underline{d}_2^{(k,i)}, \ldots, \underline{d}_N^{(k,i)})^{\mathrm{T}},\ i = 1,\ 2,\ k = 1,\ldots,K. \tag{4-1}$$

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 (4-1) of a data block has the symbol duration $T_S$.

The data modulation is either QPSK or 16QAM. In the case of QPSK modulation the data symbols $\underline{d}_n^{(k,i)}$ are generated from 2 interleaved and encoded data bits

$$b_{l,n}^{(k,i)} \in \{0,1\},\ \ l = 1,2,\ n = 1,\ldots,N,\ k = 1,\ldots,K,\ i = 1,\ 2, \tag{4-2}$$

using the equation

$$\begin{aligned} \mathrm{Re}\{\underline{d}_n^{(k,i)}\} &= \frac{1}{\sqrt{2}}(2b_{1,n}^{(k,i)} - 1), \\ \mathrm{Im}\{\underline{d}_n^{(k,i)}\} &= \frac{1}{\sqrt{2}}(2b_{2,n}^{(k,i)} - 1),\ n = 1,\ldots,N,\ k = 1,\ldots,K,\ i = 1,\ 2. \end{aligned} \tag{4-3}$$

Equation (4-3) corresponds to a QPSK modulation of the interleaved and encoded data bits $b_{l,n}^{(k,i)}$ of (4-2). In the case of 16QAM modulation data symbols $\underline{d}_n^{(k,i)}$ are generated from 4 interleaved and encoded data bits

$$b_{l,n}^{(k,i)} \in \{0,1\},\ \ l = 1,\ldots,4,\ n = 1,\ldots,N,\ k = 1,\ldots,K,\ i = 1,\ 2. \tag{4-4}$$

The applied signal point constellation for 16QAM is depicted in Figure 4-6. The mapping of interleaved and encoded data bits $b_{l,n}^{(k,i)}$ of (4-4) to the signal point constellation of 16QAM according to Figure 4-6 is listed in Table 4-5.

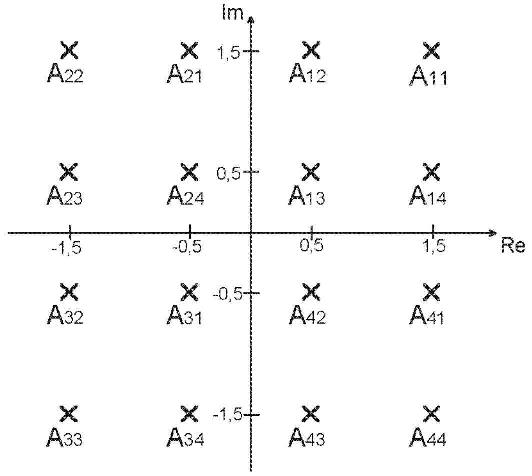



*Figure 4-6 Signal point constellation for 16QAM*

*Table 4-5 Mapping of input bits $b_{l,n}^{(k,i)}$ to the 16QAM signal point constellation*

| **Input bits $b_{l,n}^{(k,i)}$, l=1,...,4** | **Mapped on $\underline{d}_n^{(k,i)}$ =$A_{ij}$** |
|---|---|
| 0000 | $A_{13}$ = 0,5 + j 0,5 |
| 0001 | $A_{12}$ = 0,5 + j 1,5 |
| 0010 | $A_{14}$ = 1,5 + j 0,5 |
| 0011 | $A_{11}$ = 1,5 + j 1,5 |
| 0100 | $A_{42}$ = 0,5 - j 0,5 |
| 0101 | $A_{43}$ = 0,5 - j 1,5 |
| 0110 | $A_{41}$ = 1,5 - j 0,5 |
| 0111 | $A_{44}$ = 1,5 - j 1,5 |
| 1000 | $A_{24}$ = - 0,5 + j 0,5 |
| 1001 | $A_{21}$ = - 0,5 + j 1,5 |
| 1010 | $A_{23}$ = - 1,5 + j 0,5 |
| 1011 | $A_{22}$ = - 1,5 + j 1,5 |
| 1100 | $A_{31}$ = - 0,5 - j 0,5 |
| 1101 | $A_{34}$ = - 0,5 - j 1,5 |
| 1110 | $A_{32}$ = - 1,5 - j 0,5 |
| 1111 | $A_{33}$ = - 1,5 - j 1,5 |

#### *4.4.1.3 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 $\underline{c}_q^{(k)}$, q=1,...,Q, k=1,...,K, of a CDMA code $\underline{\mathbf{c}}^{(k)}$, k=1,...,K, see also Section 4.4.2.2. The impulse response of the above mentioned chip impulse filter shall be the GMSK main impulse $C_0(t)$ of duration five times the chip duration $T_C$ and time bandwidth product 0.3. The GMSK main impulse $C_0(t)$ is defined as

$$C_0(t) = \begin{cases} S(t)\prod_{i=1}^{3} S(t+iT_C), & \text{for } 0 \le t \le 5T_C \\ 0, & \text{else} \end{cases} \tag{4-5}$$

with

$$S(t) = \begin{cases} \sin\left(\pi \int_0^t g(t')\mathrm{d}t'\right), & \text{for } 0 \le t \le 4T_C \\ \sin\left(\frac{\pi}{2} - \pi \int_0^{t-4T_C} g(t')\mathrm{d}t'\right), & \text{for } 4T_C < t \le 8T_C \\ 0, & \text{else} \end{cases} \tag{4-6}$$

and

$$g(t) = \frac{1}{2T_C}\left[\mathrm{erfc}\left(2\pi \cdot 0.3 \frac{t - 5T_C/2}{T_C\sqrt{\log_e(2)}}\right) - \mathrm{erfc}\left(2\pi \cdot 0.3 \frac{t - 3T_C/2}{T_C\sqrt{\log_e(2)}}\right)\right]. \tag{4-7}$$

In equation (4-7) above

$$\mathrm{erfc}(z) = \frac{1}{\sqrt{2\pi}} \int_z^{+\infty} \mathrm{e}^{-\varsigma^2/2} \mathrm{d}\varsigma \tag{4-8}$$

denotes the complementary error function. The impulse response $C_0(t)$ according to (4-5) and the energy density spectrum $\phi_{C0}(f)$ of $C_0(t)$ are depicted in Figure 4-7.

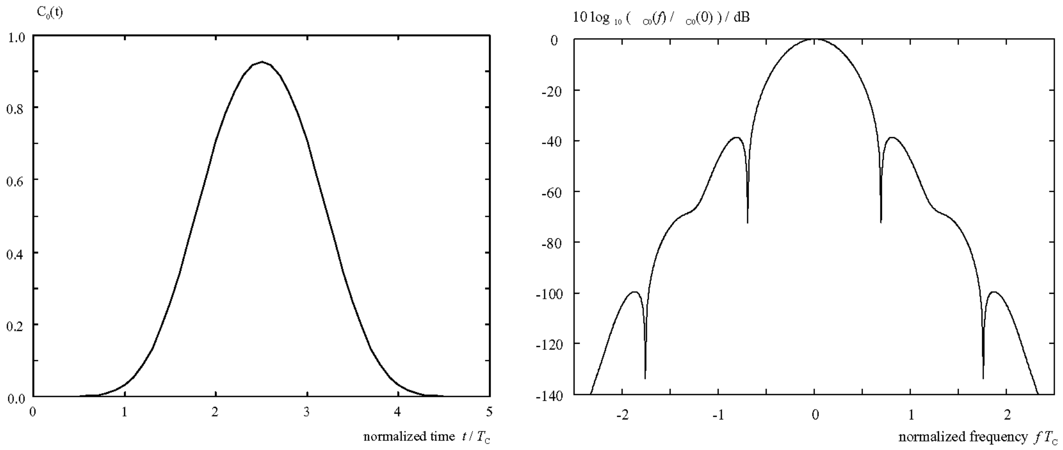



*Figure 4-7 GMSK basic impulse $C_0(t)$ and the corresponding energy density spectrum $\phi_{C0}(f)$ of $C_0(t)$*

## 4.4.2 Spreading modulation

### *4.4.2.1 Basic spreading parameters*

Each data symbol $\underline{d}_n^{(k,i)}$ of (4-1) is spread with a CDMA code $\underline{\mathbf{c}}^{(k)}$ of length

$$Q = 16. \tag{4-9}$$

Hence, the spreading factor is equal to Q according to (4-9). With Table 4-4 and (4-9) the chip duration is equal to

$$T_C = \frac{T_S}{Q} = 0.461\mu\mathrm{s}. \tag{4-10}$$

### *4.4.2.2 CDMA codes*

The elements $\underline{c}_q^{(k)}$, q=1,...,Q, k=1,...,K, of the CDMA codes $\underline{\mathbf{c}}^{(k)}$, k=1,...,K, shall be taken from the complex set

$$\underline{V}_c = \{1, j, -1, -j\}. \tag{4-11}$$

In equation (4-11) the letter j denotes the imaginary unit. The CDMA codes $\underline{\mathbf{c}}^{(k)}$ are generated from binary CDMA codes $\mathbf{a}^{(k)}$ with elements $a_q^{(k)}$, q=1,...,Q, k=1,...,K, using the relation

$$\underline{c}_q^{(k)} = (\mathrm{j})^q \cdot a_q^{(k)},\ a_q^{(k)} \in \{1,\ -1\},\ \mathrm{q} = 1, \ldots, \mathrm{Q},\ \mathrm{k} = 1, \ldots, \mathrm{K}. \tag{4-12}$$

Hence, the elements $\underline{c}_q^{(k)}$ of the CDMA codes $\underline{\mathbf{c}}^{(k)}$ are alternating real and imaginary. Table 4-6 lists binary CDMA codes which can used for $\mathbf{a}^{(k)}$, k=1,...,K, in equation (4-12). These 16 orthogonal binary CDMA codes are generated based on Walsh-Hadamard codes followed by a multiplication with a Pseudo Random (PN) sequence. Typically K is smaller than 16 and therefore in equation (4-12) less than 16 binary CDMA codes are needed. Hence, the binary CDMA codes of Table 4-6 with the smallest code numbers shall be used in practice. The CDMA codes given in Table 4-6 are one example. Other sets of 16 CDMA codes can be generated by multiplying the 16 orthogonal binary Walsh-Hadamard CDMA codes with other PN sequences. In this way, different sets of binary CDMA codes can be used in different cells.

*Table 4-6 16 Binary CDMA codes*

| | |
|---|---|
| Code 1 | (-1 -1 1 1 1 -1 1 -1 1 -1 1 -1 1 -1 -1 -1)$^T$ |
| Code 2 | (-1 -1 1 1 1 -1 1 -1 -1 1 -1 1 -1 1 1 1)$^T$ |
| Code 3 | (-1 -1 1 1 -1 1 -1 1 -1 1 -1 1 1 -1 -1 -1)$^T$ |
| Code 4 | (-1 -1 1 1 -1 1 -1 1 1 -1 1 -1 -1 1 1 1)$^T$ |
| Code 5 | (-1 -1 -1 -1 -1 1 1 -1 1 -1 -1 1 -1 1 -1 -1)$^T$ |
| Code 6 | (-1 -1 -1 -1 -1 1 1 -1 -1 1 1 -1 1 -1 1 1)$^T$ |
| Code 7 | (-1 -1 -1 -1 1 -1 -1 1 -1 1 1 -1 -1 1 -1 -1)$^T$ |
| Code 8 | (-1 -1 -1 -1 1 -1 -1 1 1 -1 -1 1 1 -1 1 1)$^T$ |
| Code 9 | (-1 1 -1 1 1 1 -1 -1 1 1 -1 -1 1 1 1 -1)$^T$ |
| Code 10 | (-1 1 -1 1 1 1 -1 -1 -1 -1 1 1 -1 -1 -1 1)$^T$ |
| Code 11 | (-1 1 -1 1 -1 -1 1 1 -1 -1 1 1 1 1 1 -1)$^T$ |
| Code 12 | (-1 1 -1 1 -1 -1 1 1 1 1 -1 -1 -1 -1 -1 1)$^T$ |
| Code 13 | (-1 1 1 -1 -1 -1 -1 -1 1 1 1 1 -1 -1 1 -1)$^T$ |
| Code 14 | (-1 1 1 -1 -1 -1 -1 -1 -1 -1 -1 -1 1 1 -1 1)$^T$ |
| Code 15 | (-1 1 1 -1 1 1 1 1 -1 -1 -1 -1 -1 -1 1 -1)$^T$ |
| Code 16 | (-1 1 1 -1 1 1 1 1 1 1 1 1 1 1 -1 1)$^T$ |

### *4.4.2.3 Spread signal of data symbols and data blocks*

With the chip impulse filter $C_0(t)$ of (4-5) the spread signal $\underline{d}_n^{(k,i)}(t)$ belonging to an arbitrary data symbol $\underline{d}_n^{(k,i)}$ can be expressed as

$$\underline{d}_n^{(k,i)}(t - T_0) = \underline{d}_n^{(k,i)} \sum_{q=1}^{Q} \underline{c}_q^{(k)} \cdot C_o(t - (q-1)T_c) = \underline{d}_n^{(k,i)} \sum_{q=1}^{Q} (\mathrm{j})^q \cdot a_q^{(k)} \cdot C_o(t - (q-1)T_c). \tag{4-13}$$

In equation (4-13) $T_0$ denotes an arbitrary time shift. The transmitted signal belonging to the data block $\underline{\mathbf{d}}^{(k,1)}$ of (4-1) transmitted before the midamble is

$$\underline{d}_n^{(k,1)}(t) = \sum_{n=1}^{N} \underline{d}_n^{(k,1)} \sum_{q=1}^{Q} \underline{c}_q^{(k)} \cdot C_o(t-(q-1)T_c - nT_c) \tag{4-14}$$

and for the data block $\underline{\mathbf{d}}^{(k,2)}$ of (4-1) transmitted after the midamble

$$\underline{d}_n^{(k,2)}(t) = \sum_{n=1}^{N} \underline{d}_n^{(k,2)} \sum_{q=1}^{Q} \underline{c}_q^{(k)} \cdot C_0(t-(q-1)T_C - nT_C - NQT_c - L_m T_c). \tag{4-15}$$

### 4.4.3 Training sequences for spread bursts

Section 4.3.1 contains a description of the spread speech/data bursts. The spread speech/data bursts contain $L_m$ midamble chips which 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

$$\underline{V}_m = \{1,\ j,\ -1,\ -j\}. \tag{4-16}$$

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

$$\underline{m}_i^{(k)} = (j)^i \cdot m_i^{(k)},\ m_i^{(k)} \in \{1,\ -1\},\ i = 1,\ldots,L_m,\ k = 1,\ldots,K. \tag{4-17}$$

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 $m_i^{(k)}$, $i = 1,\ldots,L_m$, $k = 1,\ldots,K$, of (4-17) for the complex midambles $\underline{\mathbf{m}}^{(k)}$, $k$=1,...,$K$, of the $K$ users are generated according to Steiner's method [2] from a single periodic basic code

$$\mathbf{m} = \left(m_1, m_2, \ldots, m_{L_m+(K-1)W}\right)^{\mathrm{T}},\ m_i \in \{1,\ -1\},\ i = 1,\ldots,(L_m + (K-1)W). \tag{4-18}$$

The elements $m_i$, $i = 1,\ldots,(L_m + (K-1)W)$, of (4-18) fulfill the relation

$$m_i = m_{i-P},\ i = (P+1),\ldots,(L_m + (K-1)W). \tag{4-19}$$

The P elements $m_i$, $i = 1,\ldots,P$, of one period of $\mathbf{m}$ according to (4-18) are contained in the vector

$$\mathbf{m}_{\mathrm{P}} = \left(m_1, m_2, \ldots, m_P\right)^{\mathrm{T}}. \tag{4-20}$$

With $\mathbf{m}$ according to (4-18) the $L_m$ binary elements $m_i^{(k)}$, $i = 1,\ldots,L_m$, $k = 1,\ldots,K$, of (4-17) for the midambles of the $K$ users are generated based on Steiner's formula

$$m_i^{(k)} = m_{i+(K-k)W},\ i = 1,\ldots,L_m,\ k = 1,\ldots,K. \tag{4-21}$$

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 $\mathbf{m}_{\mathrm{P}}$ according (4-20).

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 $\mathbf{m}_{\mathrm{P}}$ according to (4-20).

In the following several periods $\mathbf{m}_{\mathrm{P}}$ according (4-20) 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 4-7 always 4 binary elements $m_i$ are mapped on a single hexagonal digit.

*Table 4-7 Mapping of 4 binary elements $m_i$ on a single hexagonal digits*

| **4 binary elements $m_i$** | **mapped on hexagonal 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 $\mathbf{m}_P$ according to (4-20) and hence of the specified midamble code sets $\underline{\mathbf{m}}^{(k)}$, $k$=1,...,$K$, will be also given.

The spread speech/data burst 1 described in Section 4.3.1 contains $L_m$ equal to 296 midamble chips and can be used for different cases that are given in Table 4-8. In case 1.1, K equals 8, W equals 33, and P equals 264. Note: *In case 1.4, some symbols of the data symbol field following the midamble might have to be punctured in order to increase the guard interval.*

*Table* 4-9 contains periods $\mathbf{m}_P$ according (4-20) for case 1.1.

*Table 4-8 Summary of the cases for which the different burst types and training sequences can be used.*

| Case number | Spread speech/ data burst number | Link direction | Number of different users per time slot | Channel time dispersion | Training sequences given in Table |
|---|---|---|---|---|---|
| 1.1 | 1 | uplink | up to 8 | up to 15 µs | Table 4-9 |
| 1.2 | 1 | uplink | up to 4 | up to 25.5 µs | Table 4-10 |
| 1.3 | 1 | downlink | independent; for all numbers | up to 25.5 µs | Table 4-10 |
| 1.4 | 1 | uplink | all bursts in the same time slot allocated to one and the same user | up to 53.5 µs | Table 4-11 |
| | | downlink | independent; for all numbers | | |
| 2.1 | 2 | uplink | up to 8 | up to 5.5 µs | Table 4-12 |
| 2.2 | 2 | uplink | all bursts in the same time slot allocated to one and the same user | up to 15 µs | Table 4-13 |
| | | downlink | independent; for all numbers | | |

*Note: In case 1.4, some symbols of the data symbol field following the midamble might have to be punctured in order to increase the guard interval.*

*Table 4-9 Periods $\mathbf{m}_P$ according (4-20) for case 1.1 (confer Table 4-8).*

| Periods $\mathbf{m}_P$ of length $P$=264 | Degradation in dB, ([2], equation (38)) |
|---|---|
| B257 133D 209C B1E3 D538 80A1 3ACC 53EB 12C6 D826 1547 0344 85FF 5BA0 F4CD F495 73 | 0.6859 |
| B635 B3B3 4056 AEED E2AD 3797 1F00 6603 A8D3 87A9 EEED 0B8C 0241 A920 1ED4 2306 75 | 0.9418 |
| 89A7 3444 A1D4 210C C049 51F3 FF92 7107 5962 0993 85A9 5EB4 BE7C 3B81 47F7 3D94 CA | 0.8557 |
| E5E2 760D 86D0 C3C7 1BC8 E95F 70C6 91C8 8089 274D 0680 3FF9 58FD 92A8 4D4A 22BA CE | 0.9151 |
| 0F02 97EA FBD4 1D57 A464 310C 9DAA 11D9 2981 903A EC7C 8924 BC14 77B7 7858 C6C1 94 | 0.8418 |
| 40A3 F18E 46AD 3640 C074 822A B3B3 0ACB 5938 FE32 DE16 58A6 9141 3953 F281 1860 EF | 0.8734 |
| 20E8 34F0 14D6 6F40 722B 5BFF 4E30 FCC4 8319 C883 1176 FCAA 4BD2 8526 8678 A236 98 | 0.8950 |
| DC3D 5503 7D2E 907F 3329 C511 CB81 880E BD06 4700 18CF D7CF 097A 4889 29CB 4792 72 | 0.7378 |

In case 1.2, K equals 4, W equals 57, and P equals 240. In case 1.3, K equals 1, W equals 57, and P equals 240. Table 4-10 contains periods $\mathbf{m}_P$ according (4-20) for case 1.2, and case 1.3.

*Table 4-10 Periods $\mathbf{m}_P$ according (4-20) for case 1.2, and case 1.3 (confer Table 4-8).*

| Periods of length $P$=240 | Degradation in dB |
|---|---|
| 4976 A5B5 1842 319F CB04 7165 6C2F 4864 2E16 031C B44C FABF 66E6 85EE | 0.8727 |

| | |
|---|---|
| 1157 | |
| FCC6 F0AB 7C19 06B3 416B C630 9884 ACC5 64D1 97A0 BED7 C38C 5EE9 9B60 20AA | 0.7906 |
| 17F0 C80A 15DC 3080 64C1 C4F0 757B 4C7B B8E4 9474 4DCE F593 6894 8B82 337A | 0.9248 |
| A81B 226A 8711 0E90 F8A1 1A0F DAD1 AF86 F646 030D BC85 5FE4 835A DC73 D858 | 0.7545 |
| 040C D19E 7323 5E86 4C3F 90C7 D55D AC4A 2CCF 13C0 EEC2 416A 7BCB 9410 281F | 0.9321 |
| 0B14 545B 86A1 30EC 8422 07F5 A7A6 CFA2 7633 D50F 4B86 3FC0 64D0 8133 D971 | 0.6956 |
| 2DFA 5B21 ABC4 CFE1 C721 BC28 92A3 C62F E04D 5B01 C1A2 0843 5F33 220F B72A | 0.8549 |
| 8010 9103 DF94 AADC 30EC 0928 0E67 E75C 9D05 9F0C B4FE 4510 56B2 CD41 3E18 | 0.9306 |

In case 1.4, K equals 1, W equals 117, and P equals 180. Table 4-11 contains periods $\mathbf{m}_P$ according (2-29) for case 1.4.

*Table 4-11 Periods $\mathbf{m}_P$ according (2-29) for and case 1.4 (confer Table 4-8).*

| Periods of length $P$=180 | Degradation in dB |
|---|---|
| 058B 8AFD FE2C D161 077A 1DC1 D671 C6B3 2044 C5A5 809C D | 0.7695 |
| 100C 5454 2A87 0947 7921 F46F C192 65B3 8529 289F 9B2B 7 | 0.6636 |
| C267 E0ED 37AA B8B4 037F 8527 1A39 60B5 4CFB 1A20 8424 E | 0.8443 |
| EF0F 9477 0104 E4A4 37A2 316A 78E0 93FD EB9A 3112 0993 2 | 0.9377 |
| C3F3 0E95 EAE9 8119 FB2D C6C5 8597 44F7 0938 2584 4152 4 | 0.8699 |
| D027 D317 C521 9C59 664C BD8F C129 17B7 D7A2 A318 01A5 8 | 0.7816 |
| 0A44 9557 AF3F 7095 7762 C473 23F9 8678 2419 0DE1 E025 A | 0.9472 |
| 8656 5F14 95C9 AC17 BD83 8E99 83E2 1444 533E 038D BB50 1 | 0.9198 |

The spread speech/data burst 2 described in section 4.3.1 contains $L_m$ equal to 107 midamble chips and can be used for different cases given in Table 4-8. In case 2.1, K equals 8, W equals 12, and P equals 96. Table 4-12 contains periods $\mathbf{m}_P$ according (4-20) for case 2.1.

*Table 4-12 Periods $\mathbf{m}_P$ according (4-20) for case 2.1 (confer Table 4-8).*

| Periods of length $P$=96 | Degradation in dB |
|---|---|
| 48A7 4F42 2A78 3A80 1CB6 736E | 0.6842 |
| C4C0 03AA 09A1 FADC D462 9E62 | 0.8320 |
| 5821 EBEE 07A6 91C0 8929 4CC1 | 0.6881 |
| 98CD 3057 C349 3F57 9686 810A | 0.9026 |
| 9440 AF0C 9BD0 386A B9B6 13BC | 0.7971 |

| | |
|---|---|
| B5F5 24D0 3BE3 0682 A118 89A2 | 0.7828 |
| 52C4 9D1C 9C41 6588 30AC F43F | 0.9496 |
| 1F4A A362 484D F488 04E3 2BE3 | 0.7626 |

In case 2.2, K equals 1, W equals 32, and P equals 76. Table 4-13 contains periods $\mathbf{m}_{\mathrm{P}}$ according (4-20) for case 2.2.

*Table 4-13 Periods $\mathbf{m}_{\mathrm{P}}$ according (2-29) case 2.2 (confer Table 4-8).*

| Periods of length $P$=76 | Degradation in dB |
|---|---|
| 3731 7058 77C9 1EA2 414 | 0.5825 |
| A599 C7C8 69D1 5F25 002 | 0.9177 |
| 88E9 A25E F158 0A48 C38 | 0.7769 |
| 88E9 A25E F158 0A48 C38 | 0.7769 |
| 0E95 0137 90D1 172E 6B7 | 0.6832 |
| EE96 C227 8186 3952 07E | 0.7473 |
| E05E 99A5 5D38 1849 0DE | 0.9156 |
| 88E9 A25E F158 0A48 C38 | 0.7769 |

In the case of the downlink, $2K$ 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, $2K'$ 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 $2K$ (or $2K'$) 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.

### 4.5 Examples of gross bit rates and service mappings

This chapter presents the gross bit rates of WB-TDMA/CDMA and some examples how the burst can be used to provide different data rates.

*Table 4-14 Gross bit rates of different burst types of WB-TDMA/CDMA*

| **Burst type** | **Modulation** | **Gross bit rate per single slot (kbit/s)** | **Total gross bit rate (using all slots) (Mbit/s)** |
|---|---|---|---|
| Spread Speech/Data 1 | QPSK | 24.3 | 1.55 |
| Spread Speech/Data 1 | 16QAM | 48.6 | 3.11 |
| Spread Speech/Data 2 | QPSK | 29.5 | 1.89 |
| Spread Speech/Data 2 | 16QAM | 59.0 | 3.77 |

*Note In gross bit rates per slot, no overhead due to possible idle slots or associated control channels is included.*

A service requiring a certain bit rate can be accomplished by using a combination of time slots and codes. In Table 4-15 the user bit rates of specific interest are listed. Further, examples of how these rates could be mapped onto time and code slots in the WB-TDMA/CDMA case are also given.

*Table 4-15 Examples of service mappings for WB-TDMA/CDMA.*

| required user bit rate (kbit/s) | code rate | burst type | modulation | number of basic physical channels (code/time slot) per frame |
|---|---|---|---|---|
| *8* | *0.66*<br>*0.54* | *spread speech/data burst 1*<br>*spread speech/data burst 2* | *QPSK* | *0.5* |
| *64* | *0.66*<br>*0.54* | *spread speech/data burst 1*<br>*spread speech/data burst 2* | *QPSK* | *4* |
| *144* | *0.66*<br>*0.54* | *spread speech/data burst 1*<br>*spread speech/data burst 2* | *QPSK* | *9* |
| *384* | *0.66*<br>*0.54* | *spread speech/data burst 1*<br>*spread speech/data burst 2* | *QPSK* | *24* |
| *1024* | *0.66*<br>*0.54* | *spread speech/data burst 1*<br>*spread speech/data burst 2* | *QPSK* | *64* |
| *2048* | *0.66*<br>*0.54* | *spread speech/data burst 1*<br>*spread speech/data burst 2* | *16QAM* | *64* |

## 5 Resource allocation, variable data rates

The following figure represents the layer structure and the protocols and algorithms of the UMTS TD/CDMA radio interface. Algorithms are represented in dashed boxes.

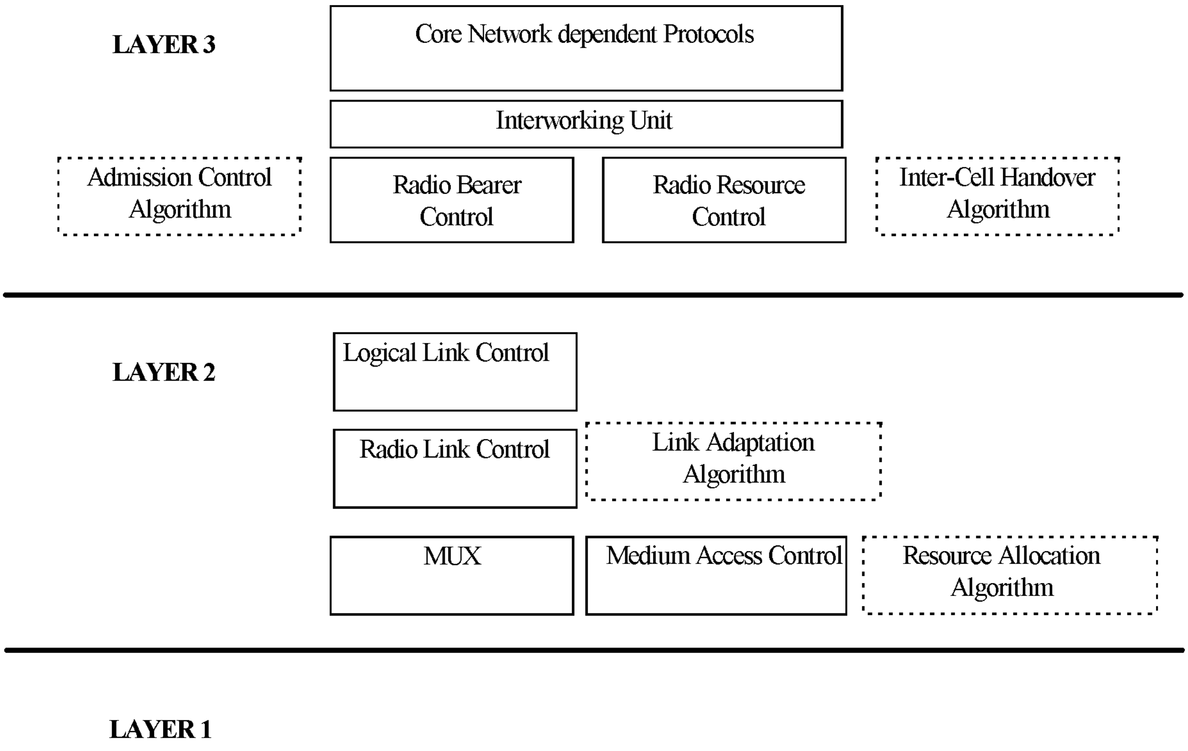



*Figure -15-1 UMTS TD/CDMA Radio interface*

### 5.1 The RLC/MAC sub-layer

RLC functions of the RLC/MAC protocol are bearer specific and RLC entities for each bearer are created during bearer setup. The RLC entity is unidirectional, so that an unidirectional bearer is represented by 2 RLC entities, one on the network side and one in the MS, while a bi-directional bearer is represented by 4 RLC entities, 2 on the network side and 2 in the MS. These entities deal with link adaptation (both traffic and radio condition adaptation).

The MAC entity is common to all bearers in a cell, i.e. all RLC functions are served by a single MAC protocol. The MAC protocol locates in the BS the MAC entity and in the MS the MAC entity.

There are two MAC/ RLC operating modes, one called Real Time (RT) and the other called Non Real Time (NRT). The former is used for radio bearers which have severe delay variation constraints and the quality is mainly fulfilled by forward error correction and power control. The latter is used for radio bearers with relaxed delay variations that allow use of backward error correction. The resource allocation and release mechanisms for the two operating modes are described in the following.

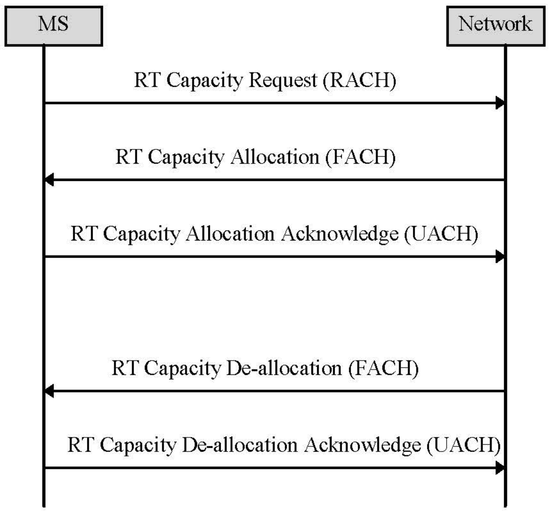
Key benefits of the developed RLC/MAC protocol are

- fast allocation and release of resources -> efficient access to radio resources
- optimized co-existence of RT and NRT operating modes
- supports an efficient Type II Hybrid ARQ scheme for NRT services.

### *5.2 RT operating mode*

Capacity for RT services is allocated in a circuit switched manner i.e. the capacity is allocated for a bearer until a specific release procedure is executed. The MAC resource allocation function can administer resources in the RT mode in a number of ways: for example, at set-up it could allocate the bearer the maximum resources that the call requires, and the call retains exclusive use of those resources throughout its duration even if only a fraction are used for most of the time. Alternatively, the maximum resources can be allocated but those which are not used can be released for use by NRT services when not required. A delay will result whilst allocated resources are recovered but blocking will not occur. Thirdly, resources need only be allocated when required and when released are added to the general resource pool. The RT signalling procedures of the proposed scheme are capable of supporting all these options.

#### 5.2.1 Real Time Transmission




For dynamic allocation, whenever a BSS RLC detects that its radio bearer needs more resources than it currently has, it requests resources from MAC. BS MAC sends a Capacity Allocation (CA) message to the MS MAC. The CA message is transmitted on the downlink MAC signalling channel (FACH) and contains the MS and bearer identifiers and the parameters (slot-code set(s)) that identify the additional channel. The CA message is acknowledged by the MS with a Capacity Allocation Acknowledgement (CAA) message which is transmitted on the random access channel for MAC related signalling (N-RACH) or a channel temporarily assigned to the MS for the acknowledgement (UACH).

Whenever a BSS RLC detects that the bearer has too many resources allocated, RLC can request MAC to decrease its resources. A Capacity De-allocation (CD) message is transmitted to the MS on the FACH and is acknowledged by a Capacity De-allocation Acknowledgement (CDA) sent on the N-RACH or UACH.

The resource allocation de-allocation procedure is the same for MS initiated changes, but the CA-CAA message exchange is initiated by the MS MAC transmitting a Capacity Request (CR) message on the N-RACH indicating the revised capacity requirements. Each of the above messages may also be transmitted on an ACCH or SDCCH should the channel be available.

### *5.3 NRT operating mode*

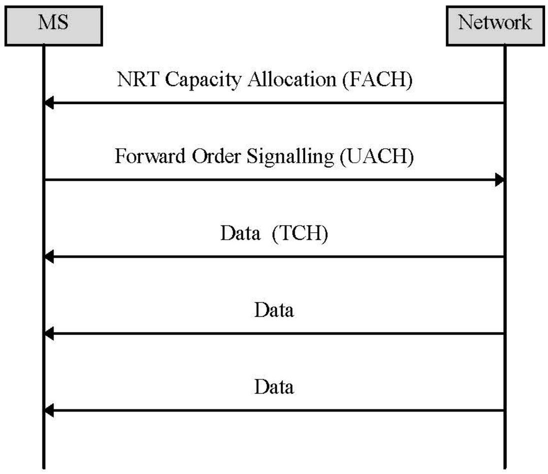
Capacity allocation for an NRT bearer is made for relatively short periods of time and the resource is automatically released at the end of the allocation period. Two NRT schemes are supported, the resource allocation scheme decides which one is the most appropriate for each case.

In the first scheme, referred to as Scheduled Allocation, type II ARQ and/or soft combining is used. Consequently data to be transmitted is scheduled by the forward order principle i.e. the receiving side identifies what data is to be transmitted in each allocation of resources.

In the second scheme, referred to as Immediate Allocation, use is made of type I selective retransmission ARQ. In this case the transmitting end selects what is to be sent and the receiving end acknowledges, requesting retransmission of what has been missed.

In both cases the BSS MAC makes allocations of NRT resources to MS for uplink or downlink data transmission. These allocations are made for relatively short periods of time specified by the MAC resource allocation process at the time of allocation. Allocations take the form of sets of slots/codes. Each set, over an interleaving period, represents an ARQ retransmission unit. (data unit). For most of the implementation that are being considered a retransmission unit is also the same as a FEC coding block. The allocation period is equivalent to one or more interleaving periods.

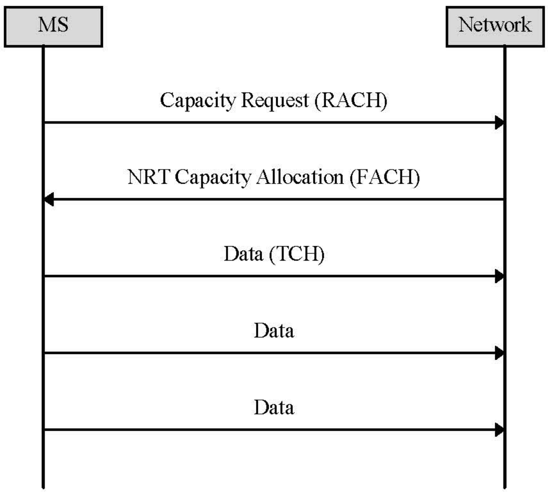
### 5.3.1 Scheduled Allocation Down-link transmission




Whenever a down link NRT radio bearer BSS RLC entity needs to transfer data, it indicates the amount of data to the BSS MAC entity. The MAC will make one or more NRT resource allocations to transfer this data. The allocations will be signalled to the destination MS MAC by Capacity Allocation (CA) messages transmitted on the FACH. The CA messages will contain the slot/code sets, interleaving period and allocation period for the data transfer together with a similar, but smaller, resource allocation for a UACH channel to be used by the MS for forward order signalling.

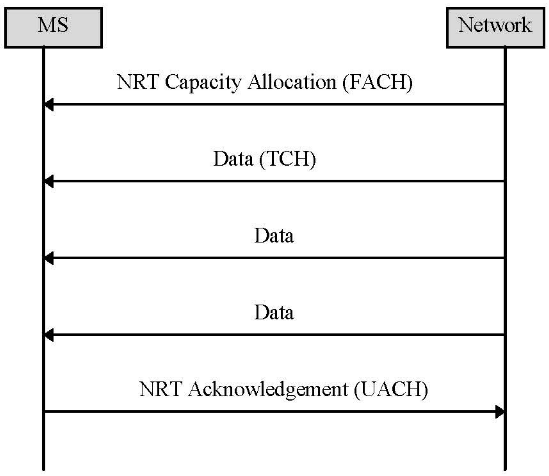
The MS will identify the data units that are to be transmitted (or re-transmitted for soft or type II combining) in the Forward Order message sent on a UACH after which the BSS will transmit the data to the MS.

### 5.3.2 Scheduled Allocation, Up-link transmission




Whenever the MS RLC entity of an up-link radio bearer needs to transfer data, it indicates the quantity of data to the MS MAC entity. The MAC then sends a Capacity Request (CR) message to the BSS MAC using the RACH signalling channel. The BSS MAC will respond with one or more Capacity Allocation messages transmitted on the FACH. In addition to the allocation of capacity for data transfer, the capacity allocation will indicate which data units are to be transmitted (or re-transmitted for soft or type II combining). The MS will respond to each allocation by transmitting the requested data units in the allocated slots using the allocated codes.

### 5.3.3 Immediate Allocation




Whether the Immediate Allocation is initiated by a BS-RLC request or after reception of an MS-MAC NRT Capacity Request, the procedure is almost the same. The BS-MAC sends a Capacity Allocation message to the MS indicating the slot/code sets, interleaving period and allocation time for data transfer. In the case of down-link data transfer the allocation will also specify UACH resources for acknowledgement signalling. Then the transmitting side transmits the data in the specified capacity. The receiving end sends an NRT Acknowledgement message when the allocation period is complete, in the uplink direction on the UACH and in the downlink direction on the FACH channel.

# 6 Radio Resource Management

## 6.1 General

For WB-TDMA/CDMA the radio resource management concept covers the following topics:

- Channel allocation
- Link Quality control
- Power control
- Admission Control
- Handover

Since UMTS requires the support of various services (in the range from 8 kbps up to 2 Mbps, RT and NRT; symmetric and asymmetric) in various environments the radio resource management concept being developed for WB-TDMA/CDMA must be capable to fulfil those key requirements. Additionally, constraints arising from the MS capabilities have to be taken into account.

### 6.1.1 Supported services and environments

According to [1] the following key requirements must be fulfilled regarding supported services in specific environments:

- Rural Outdoor: at least 144 kbps, maximum speed: 500 km/h
- Suburban Outdoor: at least 384 kbps, maximum speed: 120 km/h
- Indoor/Low range outdoor: at least 2 Mbps, maximum speed: 10 km/h
- Negotiation of bearer service attributes (bearer type, bit rate, delay, BER, up/down link symmetry, protection including none or unequal protection)
- Parallel bearer services (service mix), real-time (RT) / non-real-time (NRT) communication modes
- Circuit switched and packet oriented bearers
- Supports scheduling (and pre-emption) of bearers (including control bearers) according to priority
- Adaptivity of link to quality, traffic and network load, and radio conditions (in order to optimise the link in different environments).
- Wide range of bit rates should be supported with sufficient granularity
- Variable bit rate real time capabilities should be provided.
- Bearer services appropriate for speech shall be provided.

### 6.1.2 Mobile station classes

The number of the mobile classes has to be limited for practical reasons, avoiding the complexity actually reached for example in the GPRS. Since technical, economical and marketing issues will be the dominating factors determining the number of different mobile station classes it is difficult find a realistic scenario.

However, from radio resource management point of view the following mobile station classes are assumed as realistic:

- **simple speech MS:** half duplex operation within any timeslot; single timeslot both for UL and DL; half duplex multicode (DL: multicode; UL: single code)
  This MS provides similar capabilities as a basic GSM MS for both RT and NRT services.

- **simple MS:** half duplex operation within any timeslot; single timeslot both for UL and DL; duplex multicode (DL: multicode; UL: multicode)
  This MS is capable to support RT speech and NRT 'best effort' services up to a data rate of $\leq$ 144 kbps

- **enhanced MS:** full duplex operation within any timeslot; capability to retune frequency every timeslot;
  multislot capability; duplex multicode (DL: multicode; UL: multicode)
  This MS is capable to support all services up to the maximum data rate of 2 Mbps (RT and NRT).

Additionally, if the market demands a MS class with service capabilities situated between the simple MS and enhanced MS, the following MS class can be introduced:

- **medium MS:** half duplex operation within any timeslot; capability to retune frequency every timeslot; restricted multislot capability (UL + DL ≤ 5 TS); duplex multicode (DL: multicode; UL: multicode)

It is emphasised that regardless from the actual MS class each MS is capable to perform joint detection of up to 12 different codes within one timeslot to ensure that even the ‘simple speech MS’ is capable to detect the allocated codes in the DL. Therefore, the MS capabilities have mainly an impact on the multislot operation capability which is similar to all TDMA based systems.

Additionally, the MS may have the following capabilities:

- dual mode (WB-TDMA/CDMA - GSM)
- antenna diversity

## *6.2 Channel allocation*

For WB-TDMA/CDMA, a physical channel is characterised by its frequency bin, time slot, and spreading code as explained in the chapter on the physical channel structure

Channel allocation covers both:

- channel allocation to cells
- channel allocation to bearer services (fast channel allocation)

### 6.2.1 Channel allocation to cells

By default, separation of channels between cells is performed in the frequency and code domain, i.e.:

- carrier frequencies are allocated to one cell. Using FDD, a separate bandwidth is allocated for up- and downlink.
- A set of 16 orthogonal Walsh-codes is assigned to each cell multiplied with a cell-specific scrambling sequence of length 16. This code set is used in all timeslots of all carriers within the cell. Since this results in a sufficient number of available code sets a fixed allocation scheme can be applied resulting in a static allocation of one code set to a cell.

For initial deployment of WB-TDMA/CDMA fixed channel allocation is required.

WB-TDMA/CDMA doesn’t prevent the use of dynamic channel allocation techniques. Separation can be made both in the frequency and time domain. The type of dynamic channel allocation technique and the achievable gain is for further study.

### 6.2.2 Fast channel allocation

Fast channel allocation refers to the allocation of one or multiple physical channels to any bearer service. All bearer services capabilities for UMTS outlined in [1] are supported for the listed propagation environments (pico-, micro-, and macro-cells).

The following principles hold for fast channel allocation:

1. The basic resource unit (RU) used for channel allocation is one code / timeslot / frequency
2. Multirate services are achieved by pooling of resource units. This can be made both in the code domain (pooling of multiple codes within one timeslot = **multicode** operation) and time domain (pooling of multiple timeslots within one frame = **multislot** operation). Additionally, any combination of both is possible. For the ‘enhanced MS’, any arbitrary distribution of the allocated

resources in the code / time domain on one carrier is possible. For the simple MS, only the time domain is restricted to one timeslot.

3. For UL, at maximum 8 different users per timeslot are allowed; for DL, at maximum 12 different users are possible
4. For UL and DL, up to 12 codes can be active in one timeslot simultaneously. However, at maximum 8 codes per timeslot can be allocated to one user.
5. Every MS belonging to any MS class is capable to perform in the DL joint detection of up to 12 codes per timeslot. In the UL, except of the 'simple speech MS' each other has the full multicode capability. Therefore, the MS capabilities have only an impact on the multislot allocation which is common for all TDMA-based systems. As a consequence from the MS capabilities point of view, the channel allocation is restricted in the time domain only, whereas in the code domain the full flexibility is provided.
6. Channel allocation discriminates both RT and NRT bearer services:
    - RT services: Channels remain allocated for the whole duration the bearer service is established ('circuit-oriented'). The allocated channels may change because of a channel reallocation procedure.
    - NRT services: Channels are allocated for the period of the transmission of a dedicated data packet only ('packet-oriented'). 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.
7. Mixed allocation of RT and NRT services is possible. The channel allocation scheme takes the different C/I requirements of RT and NRT services into account by maintaining resource pools for RT and NRT services. The partitioning of the resource pools is dynamic and gives the operator the flexibility to optimise the system regarding a preferred service profile.
8. For symmetric RT services a time shift between DL and UL of 3 timeslots is made (as for GSM ) thus preventing the need of a duplex operation for the simple MS.
9. In case of asymmetric RT bearer services for the 'enhanced MS' channel allocation is made for uplink and downlink independently. For NRT services, channel allocation for up- and downlink is made independently for all MS classes.
10. In case of DTX for speech services, the resources remain allocated during the silent period although no signal is transmitted. Therefore, DTX operation improves the interference (inter- and intracell) level.
11. To utilise the mandatory multicode capability of all MS classes the channel allocation scheme prefers the multicode option, i.e. the number of allocated timeslots for any bearer service is minimised by allocating as many codes per timeslot as possible. The number of codes used by one bearer service within one timeslot is a power of 2, i.e. 1,2,4 or 8
12. 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.
13. The bursty nature of NRT services and the channel allocation according 'best effort' achieves high statistical gain thus avoiding the need of the reshuffling procedure for multi-rate NRT services
14. The required effort for channel reshuffling is reasonable because:
    - the expected long duration of multi-rate RT bearer services (e.g. video conference) lowers the expected rate of channel reshuffling. Further, due to their high resource consumption the number of multi-rate RT bearer services is limited by admission control
    - number of allocated codes per timeslot per bearer service are powers of 2, thus decreasing the reshuffling rate if timeslots are partly filled with used codes

### *6.3 Link Quality Control*

Link Quality Control is in charge to provide measures to maintain the bearer service QoS parameters taking the radio environment condition (i.e., interference level) into account.

For WB-TDMA/CDMA this implies the selection of the following transmission parameters according the current radio environment condition:

- channel coding
- interleaving depth
- burst type
- modulation

The selection of the transmission parameters and the required data rate of the bearer service directly determines the required number of resource units. In case of bearer services requiring multiple resource units the selection of the transmission parameters is equal for all allocated resources.

Some of the parameters are dependent from the system deployment and the required bearer service and remain static:

1. The burst type is cell dependent (i.e., all MS within the cell use the same burst type) and is determined during system deployment.
2. 16QAM modulation is used for 2 Mbps services only; the remaining services use QPSK. Nevertheless, a mix of QPSK and 16QAM within one cell is possible. The higher C/I requirements of 16QAM are maintained by link quality control.
3. For RT (speech, LCD, LDD) services, the interleaving depth is determined during call set-up and remains fixed for the whole call duration. To prevent long time intervals for channel allocation, the interleaving depth should not exceed 16 TDMA frames.

Since the radio environment conditions vary in time and space, link quality control has to take this into account when maintaining the bearer service QoS. The basic measure for maintaining the link quality is using power control.

Additionally, dependent from the type of bearer service the following additional measures are provided:

- for multi rate RT services, link adaptation can be performed
- for NRT services, an ARQ mechanism is used

Link adaptation for RT services means the adaptation of the channel coding according to a quality criteria derived from the C/I values of all received bursts over each interleaving period. The measurements of one bearer are averaged over all used codes, timeslots and frequencies (if frequency hopping is applied). Since the adaptation of the channel coding results in an increase or decrease of the number allocated resources the use of link adaptation for multi rate RT bearer services results in additional capacity gain since it allows to adapt the resource requirements on the current interference level. Link adaptation can be performed after each interleaving period. If additional resources are allocated (or released in case of less channel coding) the number of these resources is always 1, 2, 4, 8 or a higher power of 2 to avoid resource fragmentation.

For NRT services, an ARQ mechanism is used instead of link adaptation. Two ARQ types are foreseen:

- ARQ type 1: selective re-transmission of the disturbed data block
- ARQ type 2: ARQ mode uses hybrid coding, soft combining and reception quality estimates

The first option is being preferred for a small packets whereas the second is preferred for high rate NRT services transmitting large packets. Since the ARQ scheme is basically a data link layer protocol feature, please refer to chapter 5.

### *6.4 Power Control*

Power control is applied for WB-TDMA/CDMA to limit the interference level within the system thus reducing the intercell 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:

| | **Uplink** | **Downlink** |
|---|---|---|
| **Dynamic range** | 80 dB | 30 dB |
| **Power control rate** | variable; 2-200 cycles / second | variable; 2-200 cycles / second |
| **Step size** | 2 dB | 2 dB |
| **Remarks** | None | 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, a closed loop power control is used
- For NRT services, an open loop power control is used
- The initial power value is based on the pathloss 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
- Introduction of fast power control

### *6.5 Admission Control*

Admission Control decides whether a user respectively a bearer service requested by a user is to be admitted into the system during bearer setup, bearer re-negotiation or handover decision procedure. Due to a the different bearer services to be supported and the limited available bandwidth admission control is vital for a successful system deployment.

The decision is at least based on the following information maintained by admission control:

- available resources in the system (cell or group of cells)
- established bearer services including their QoS attributes
- ‘system load’ defined by number of resources already occupied
- Capabilities of admitted MS

One of the paramount requirements towards admission control is to provide the operator the flexibility to optimise the system load to his specific needs, e.g. one operator focuses on a maximum number of admitted NRT users whereas another one focuses on a maximum quality for admitted RT users. This flexibility can be provided by introducing bearer priorities or bearer pricing schemes.

At least the following simple rules can be applied from admission control:

1. Within a system, the percentage of admitted realtime services is restricted to a specific 'fractional load' of the overall available capacity to provide capacity used for link adaptation and handover.
2. Concerning admission of NRT services, as long as they don't require a sustainable data rate they are admitted according to best effort

## *6.6 Handover*

### 6.6.1 General

In general sense, handover (HO) is considered as the change of physical channels (both at the radio interface and within the fixed part of the access network) allocated to a call while maintaining this call. Within the scope of this chapter HO procedures are restricted to the procedures executed at the radio interface for intercell handover i.e., handover between different cells belonging either to the same or different cell layers (HCS). The intracell handover (HO within a cell) can be defined as channel reallocation and is therefore within the scope of channel assignment (or channel allocation) procedure. Further, the procedures within the fixed part of the access network are not treated in this chapter.

#### *6.6.1.1 Key requirements*

The purpose of intercell HO is to maintain a call while the user is crossing a cell border. Also measures from Radio Resource Management (e.g. directed retry) and O&M (e.g. Pre-emption) may trigger intercell HO.

The following key requirements on intercell HO are defined in [1]:

1. Provide seamless handover between cells of one operator. Non-seamless handover should be provided when seamless handover is not feasible
2. Efficient handover between UMTS and 2nd generation systems, e.g. GSM, should be supported, if the 2nd generation systems support these bearer services.
3. The UTRA should not prevent seamless handover between different operators or access networks

#### *6.6.1.2 Assumptions*

1. If parallel bearer services are established on a single mobile terminal they are being served by the same cell. Consequently, a handover is performed for all these bearer services simultaneously.
2. For WB-TDMA/CDMA, only hard handover is mandatory. Soft handover (macro diversity) is not necessary. However, the proposal doesn't prevent the use of soft handover.
3. Since mobile assisted handover is assumed, the following requirements arise:
   - to allow path loss measurements within each cell a continuous pilot signal is transmitted with a constant transmission power
   - to allow pre-synchronisation to any cell to perform cell identification a broadcast signal containing synchronisation information is transmitted with a constant transmission power
   - within each cell broadcast information is sent also being used for the unambiguous identification of the cell.

   The proposed WB-TDMA/CDMA BCCH structure (refer to chapter 3.3.2) meets above requirements

4. Several MS classes are supported which are characterised by the number of RX/TX units, the synthesiser (s) tuning speed and the provision of duplexer units. For the enhanced MS no specific restrictions concerning handover procedures exist. However, in order to provide a handover scheme also being supported from less complex MS the following assumptions are made:
    - The simple MS is able to execute measurements at least in one time slot in the frame while it is neither receiving nor transmitting traffic
    - While the MS is being active in either WB-TDMA/CDMA or GSM mode measurements on beacon frequencies of both GSM and WB-TDMA/CDMA are possible. It is assumed that the simple MS cannot perform these measurements in parallel (simultaneously at any time instant). For more complex MS classes these restriction may not hold.
5. To allow neighbour cell identification for the simple MS 'idle' periods are provided
6. HO scheme does not mandatory require synchronised cells. However, HO between synchronised cells is an option.

#### *6.6.1.3 Discrimination NRT <-> RT bearer services*

For the HO procedure the discrimination between real-time (RT) and non-real-time (NRT) bearer services is possible due to the different requirements on the HO procedure:

- RT bearer services have stringent delay requirements but have more relaxed BER requirements. Since the HO shall be seamless , i.e. not noticeable for the user, the handover procedure should cause no extra delay. A seamless handover for RT services does not imply the need of a lossless handover.
- NRT bearer services have very low (possibly unconstrained) delay requirements but stringent BER requirements. Concerning HO of NRT services, the HO shall be lossless i.e., no data loss at expense of a possible increase of delay.

### 6.6.2 HO scheme for RT bearer services

The basic HO scheme is similar to GSM i.e., the basic scheme is a mobile assisted, network evaluated and decided, hard handover using backward signalling. Nevertheless, improvements are introduced to consider the corner effect, HCS and UMTS specific traffic / service requirements. Furthermore, to support a seamless HO for various RT services, an accelerated HO access is suggested.

Mobile evaluated handover with background network control also offers possibilities for faster HO decision and therefore is not excluded principally but left for further study, since it leads to extra downlink signalling overhead due to additional broadcast information to control the HO evaluation algorithm in the MS, and to less flexible HO evaluation algorithm selection due to the necessity of modifications in the MS.

#### *6.6.2.1 HO criteria*

HO initiation and decision shall be performed on the following non-exhaustive criteria list:

1. Receive level of serving cell and neighbour cells
2. Receive quality of traffic and /or signalling channel of serving cell
3. MS-BS distance
4. MS mobility (estimation of MS speed and direction)
5. MS power budget
6. traffic reasons (overload handling)
7. Type of bearer service
8. Cell priority

The introduction of a C/I based HO criteria is an option for further study

#### *6.6.2.2 HO Phases*

##### 6.6.2.2.1 Measurement phase

Measurements performed by MS:

1. Receive level of downlink traffic channels of serving cell and pilot channel (PICH, refer to chapter 3.3.2) of neighbour cells used for pathloss estimation. PICH measurements are performed in the period between TX and RX.
2. Receive quality of downlink traffic and signalling channels of serving cell
3. MS mobility (estimation of MS speed using time thresholds)
4. Observed time difference between serving and candidate cells for timing advance estimation (‘pseudo-synchronous’ handover)

The measurements are pre-processed and transmitted periodically to the network. Additionally, the MS can transmit measurements instantaneously on certain events (e.g., fast signal strength decay due to the corner effect) or if ordered by the network requesting extra measurement reports.

Further, measurement pre-processing allows trend analysis to detect an ‘Emergency HO condition’ resulting in an accelerated handover procedure.

Frame synchronisation and cell identification is obtained from the SCH (refer to chapter 3.3.2) which is detected in the period between TX and RX. Since the PICH provides an implicit cell identification after initial neighbour cell SCH detection the PICH is sufficient for neighbour cell identification.

Measurements performed by BS:

1. Receive quality of uplink traffic and signalling channels
2. Interference level on idle channels

Further, the BS maintains up-to-date information about the difference between its own and neighbour cells' timing (‘pseudo-synchronous’ handover) and about the load situation.

In case of urban areas, the BS maintains information concerning the estimated MS speed and direction allowing the network to prepare a handover to specific cell(s) to mitigate the corner effect (‘Emergency HO’).

##### 6.6.2.2.2 HO initiation phase

Here we distinguish between HO initiation in a single layer and between layers in HCS.

###### *6.6.2.2.2.1 HO initiation in single layer structures*

A handover is initiated if one of the following condition is fulfilled with decreasing priority:

1. ‘Emergency HO condition’, i.e. the rapid decay of signal strength due to corner effect. Detection of this condition is possible both through signal strength measurements (performed by MS) and through estimation of MS speed and direction allowing the network to prepare a handover to a specific cell (or cells).
2. Receive quality of traffic and /or signalling channel of serving cell falls below a specific threshold and cannot be mitigated by means of link quality control. Both up- and downlink shall be handled separately (e.g., by introducing weight factors) to consider asymmetric bearer services.
3. Receive level of serving cell falls below a specific threshold
4. MS-BS distance exceeds a specific threshold
5. Power budget exceeds hysteresis margin.
6. Network initiated HO due to traffic reasons (overload handling)

###### *6.6.2.2.2.2 HO initiation in HCS*

In addition to the intra layer HO initiation conditions listed above the following conditions initiate inter layer HO

- MS mobility recommends inter layer HO (e.g., microcell -> macrocell)
- Sufficient level HO, i.e. if a cell of a subordinate layer is received with sufficient RX level, an inter layer HO to this subordinate cell is initiated. Thus, as long as no other constraints hold, the traffic is concentrated in the lower layers.
- Priority level assigned to cells; the priority is service and operator dependent

##### 6.6.2.2.3 HO decision phase

In the basic scheme HO decision is performed in the network only. HO decision is based on:

- radio related criteria
- traffic and service related criteria

For the first case, from the HO candidates the target cell is selected with:

- minimum path loss
- best C/I estimate

For traffic / service related HO decision the following criteria must be considered:

1. Candidate cell supports the type of bearer service the handover decision is pending.
2. Load situation in candidate cells
3. Bearer service priority (if HO for multiple bearer services is pending and the target cell is only capable to serve a subset)

For both radio and traffic / service related decision the cell priority (in HCS environment) is considered.

##### 6.6.2.2.4 HO execution phase

As basic scheme, hard backward handover is assumed. Concerning handover access two options are foreseen:

- handover access with known timing advance (‘pseudo-synchronous’ handover), being the standard case
- handover access without known timing advance (‘asynchronous’ handover), in the (seldom) case of unknown timing relation between the old and new serving base station.

As soon as the HO decision has been made the network sends a handover command to the MS providing the new serving cell and the identities of the allocated physical channels in the new serving cell. An optional bearer re-negotiation has to take place if the new serving cell cannot support the bearer service agreed at call establishment.

As soon as the MS has received the HO command comprising the target cell, the traffic channels and the type of handover (asynchronous/pseudo-synchronous), it performs a HO access on the new serving cell:

- In case of a pseudo-synchronous handover the handover access is made on the new traffic channel since the timing advance is known. Thus, the break duration (time interval between release of old traffic channels and establishment of the new ones in the target cell) is minimised.

- In case of an asynchronous handover (which is the rare case) the handover access is performed on a dedicated random access channel to obtain time alignment from the BS before switching to the new traffic channels.

If the access to the target cell was not successful the MS returns to the old serving cell. If this is no longer possible due to rapid signal decay (e.g., corner effect) it shall be possible to make an emergency HO access to a cell which has been indicated by the network as 'Emergency HO cell'. This is an example for a forward HO to a cell (or limited number of cells) which have been prepared by the network for this accelerated handover. However, the detailed procedure of this handover scheme is for further study.

### 6.6.3 HO scheme for NRT bearer services

NRT bearer services require a lossless HO. Thus, break duration time is of minor importance whereas an ARQ protocol on higher network layers guarantees a lossless data transfer. Since the data of NRT bearer services is transferred in a packet oriented mode over the air interface similarities to GPRS arise: Handover is performed in-between the transmission of data packets. For data rates supported by GPRS a handover of a WB-TDMA/CDMA NRT bearer service to GPRS and vice versa is supported. In general, a HO for packet data services is more like a cell selection than a traditional HO.

In case a MS has allocated RT and NRT services at a time by default the handover for the RT service is prioritised over the NRT service, i.e. the NRT services follow the RT services into the new cell. Nevertheless, other HO strategies are possible, i.e. due to priority of the bearer services.

In case an MS only uses NRT services, a forward mobile evaluated (MEHO) handover with background control from the network utilising broadcast HO parameters is proposed. For enabling the network to control NRT HO immediately, we also consider a mobile assisted, network evaluated handover as option for further study.

#### *6.6.3.1 HO Criteria*

HO initiation shall be performed on the following criteria:

1. receive level of pilot channel (PICH) of serving cell and of neighbour cells
2. receive quality of traffic and/or signalling channel of serving cell based on BER or ARQ repeat counters
3. MS mobility
4. routing area hysteresis
5. network initiation due to traffic constraints

#### *6.6.3.2 HO Phases*

##### 6.6.3.2.1 Measurement phase

<u>Measurements performed by MS:</u>

1. The MS measures receive level of serving cell and neighbour cells on the pilot channel. Additionally, the MS synchronises onto and decodes the neighbour SCH's. Thus, a unique identification of candidate neighbour cells is possible during and in-between packet data transfer.
2. Receive quality of downlink traffic and signalling channels of serving cell can be measured in terms of BER, receive quality of up- and downlink traffic channels can be estimated by ARQ repeat counters.
3. MS mobility can be estimated by time thresholds

##### 6.6.3.2.2 Cell (Re-)Selection

A cell re-selection is initiated by MS if a packet transmission has ended and if one of the following conditions is fulfilled with decreasing priority:

1. receive level of pilot channel of the serving cell falls below a certain threshold

2. receive quality of the serving cell falls below a certain threshold

The selection of a new cell depends on

1. pathloss criterion fulfilled
2. MS mobility
3. cell priority
4. routing area

The HO (Cell selection resp.) is a forward type of handover, i.e. signalling channels are established in the new cell before the next data packet starts.

Optionally, the network has the possibility to command an MS to perform a cell re-selection to a specific cell, e.g. for load regulation conditions. This command overwrites MS decisions.

## *6.7 Handover in TDD mode*

If WB-TDMA/CDMA uses TDD operation the TDMA frame (8 time-slot frame) is divided between uplink and downlink transmission. WB-TDMA/CDMA supports the TDD operation with cell by cell basis configuration of the distribution between up- and downlink taking advantage of asymmetric bearer services.

The handover scheme described in the previous section is also suitable for the TDD operation. Since the TDD operation requires locally synchronised cells the MS is capable to perform neighbour cell measurements and identification. If in TDD mode the channel allocation algorithm considers to provide a sufficient measurement window for the simple MS.

### 6.7.1 Handover between FDD and TDD

Regarding the FDD/TDD handover, a MAHO can still used through the monitoring of the neighbour cells as described in the previous section. For a MS in FDD mode, and by using the idle period between reception and transmission of a burst, a MS can then always monitor a TDD pilot channel every frame provided that the TDD cell has at least 3 TS dedicated to the downlink, which should be true given the downlink biased nature of transmissions which is envisaged in TDD mode. The idle frame allows to acquire the TDD frame synchronisation. The handover itself can be optimised through the use of pseudo-synchronisation information, so that the MS knows the relative timing of the TDD cell. Otherwise an asynchronous handover is performed. The TDD to FDD handover follows similar principles and is even easier since in FDD mode the pilot channel is transmitted in all timeslots in every frame and the SCH is transmitted more frequently.

FDD/TDD handovers can then be achieved with simple MS with only one synthesiser and no duplexer. This has the potential for a seamless operation between the TDD and the FDD mode.

## *6.8 Handover between WB-TDMA/CDMA and GSM*

Compatible multiframe structures of WB-TDMA/CDMA and GSM make synchronisation of a MS to both systems feasible. Indeed, the mapping of the traffic slots in the WB-TDMA/CDMA multiframe structure (= 13 frames) is identical to the mapping in GSM. Idle frames may therefore be used to track GSM SCH and the monitoring window between transmit and receive may be used to perform GSM BCCH carrier power measurements

Therefore, efficient handover between WB-TDMA/CDMA and GSM is possible under the prerequisite that a WB-TDMA/CDMA base station would maintain a neighbour list containing also the nearest GSM neighbours and vice versa.

Thus, in case of a handover from WB-TDMA/CDMA to GSM a UMTS mobile station being active in UMTS mode could perform GSM neighbour cell measurements and could presynchronise with the strongest GSM neighbour cells for cell identification.

If in the WB-TDMA/CDMA access network the condition to perform a HO to the GSM cell is detected, in interworking with the GSM network it is decided whether a handover of the bearer service is possible (service is supported from GSM network) or not. In the latter case, a bearer re-negotiation might be performed to adapt the bearer QoS to the GSM network capabilities. If the handover to the GSM network is possible, the handover execution is performed similar to the known GSM inter-MSC HO procedure.

In case of a handover from GSM to WB-TDMA/CDMA, the UMTS mobile station being active in GSM mode uses the monitoring window between transmission and reception and the GSM 'idle' frame to perform the WB-TDMA/CDMA pilot channel measurements and uses the WB-TDMA/CDMA SCH to obtain frame synchronisation and cell identification.

If in the GSM network the condition to perform a HO to the WB-TDMA/CDMA cell is detected, in interworking with the WB-TDMA/CDMA network it is decided whether a handover of the bearer service is possible or not. If the handover to the WB-TDMA/CDMA network is possible, the handover execution is performed similar to the known GSM inter-MSC HO procedure.

**References:**

[1] ETR (04-01): 'Requirements for the UMTS Terrestrial Radio Access System', Vers. 2.6.1

[2] B. Steiner; P. Jung: Uplink channel estimation in synchronous CDMA mobile radio systems with joint detection. The fourth International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC'93), Yokohama, Japan, September 8-11, 1993.

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# Concept Group Delta
# Wideband TDMA/CDMA

## Evaluation Report - Part 2
## V 2.0 b

# Table of Contents

# 1. Introduction

In this part of the evaluation report mixed services for WB-TDMA/CDMA are considered. The report focuses on the following items:

- Service aspects for mixed services
- Variable bit rates in WB-TDMA/CDMA
- Realization of different service classes
- Visualization of mixed services

The system parameters used in this part 2 of the evaluation report are not necessarily identical with the system parameters used in the other parts of the evaluation report. Hence, all information contained in this part of the report shall only demonstrate the ability of WB-TDMA/CDMA to support mixed services and shall not be understood as a final specification.

# 2. Services aspects

## 2.1 Two important questions

In the following, the services aspects with particular respect to mixed services scenarios of WB-TDMA/CDMA are considered. First, the following two important questions must be answered:

- How can services with varying quality of service (QoS) criteria be used by various users when simultaneously maintaining a preferably high spectral efficiency?
- How can different services with different QoS criteria be used by a particular user?

Both questions must be jointly addressed when dealing with services aspects. Essentially, the two questions address the same problem, namely radio resource management. In this section, we address possible answers to these questions.

## 2.2 Standard service deployment (SSD) approach

A conservative answer is based on a fixed network planning approach which is presently deployed in second generation networks for speech and low data rate transmission. This approach is termed standard service deployment (SSD) approach. The network planning sets out from that particular service which is mostly used. This service will be termed the standard service in what follows. This service is of highest economical interest to the network operators and service providers because it will make up for the largest part of their revenue. All parameters of the standard service, including the link level parameters, are chosen in such a way that a highest possible spectral efficiency is assumed when solely using this standard service.

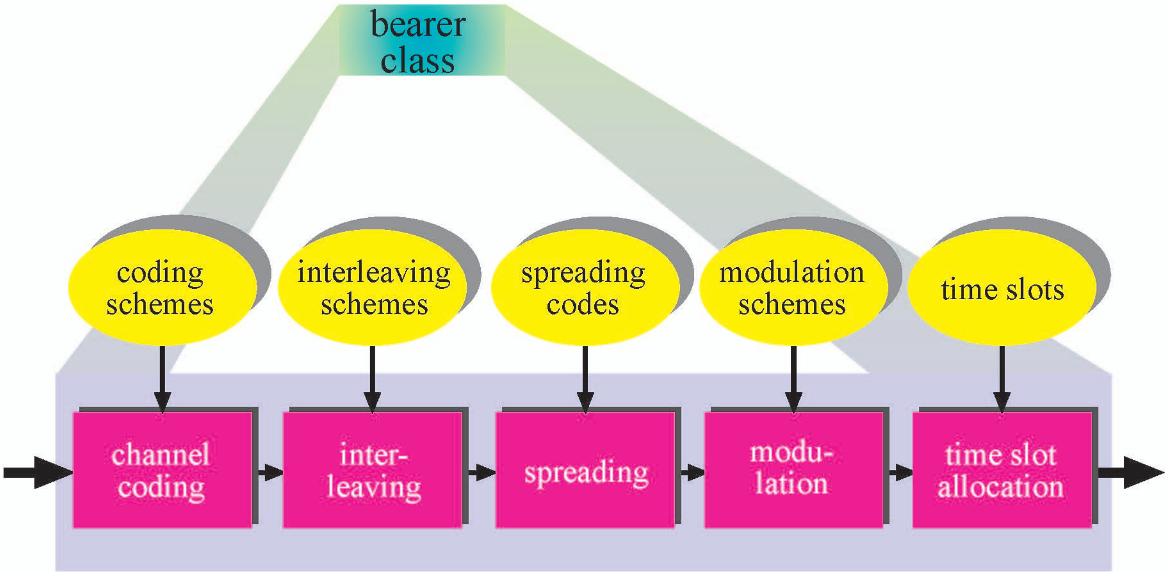


*Figure 2.1 Bearer class*

The standard service is provided by using a particular bearer class. *Figure 2.1* illustrates the link level parameters which belong to such a bearer class which can be provided by WB-TDMA/CDMA. First, the coding scheme is selected from a toolset of various coding schemes which comprises e.g. turbo codes, conventional convolutional codes, block codes such as Reed-Solomon codes etc. These coding schemes are parametrized when selected for a particular service. The selection is based on the desired QoS criterion, in particular by the desired bit error ratio or the desired packet loss ratio in the case of packet services. For instance, the code rate of the overall coding scheme, possible repetition and puncturing schemes, the constraint length of convolutional codes and the possible concatenation of schemes is set. Since the coded data are interleaved, the interleaving scheme is chosen from the interleaving schemes toolset which comprises for instance block and convolutional interleaving. Like in the case of the channel coding, the choice of the interleaving is based on the QoS criteria, in particular the delay criteria. In the case of unconstrained delay services, the choice of the interleaving is based on the desired throughput criterion. To facilitate an easy adaptation of the throughput to the time varying channel conditions, the interleaving depth can be chosen adaptively with respect to the actual correlation time of the channel. In this case, an on-the-fly optimization of the spectral efficiency is possible. Then, the spreading codes to be used in the spreading modulation are set. Afterwards, the modulation scheme to be used for data modulation, e.g. QPSK or 16QAM, is set together with the modulation scheme for spreading. The latter modulation scheme is based on binary continuous phase modulation (CPM). In WB-TDMA/CDMA GMSK is used.

A particular frequency reuse scheme with a given cluster order *r* is associated with this choice. The cluster order *r* designates the number of cells per cluster and is sometimes also called cluster size or frequency reuse factor. All other services which can now be seen as supplementary services must be deployed in the same cluster order *r*. All parameters of these supplementary services, including the link level parameters, have to be chosen to comply with the constraints defined by the deployment of the standard service. This conservative SSD approach is not optimal with respect to a joint high spectral efficiency. However, this conservative SSD approach enables second generation network operators a backward compatibility with their existing networks. Since the conservative SSD approach does not call for a flexible adaptation of the network, taking system load, environment and user requests into account, the signaling overhead required for supporting a variety of services can be kept small. WB-TDMA/CDMA is capable of supporting this conservative SSD approach.

The SSD approach allows the utilization of different services with varying QoS criteria by various users when simultaneously maintaining a rather high spectral efficiency. The rather high spectral efficiency will be achieved owing to the fact that the network planning has been done with respect to the standard service. Also, different services with different QoS criteria can be used by a particular user with high spectral efficiency.

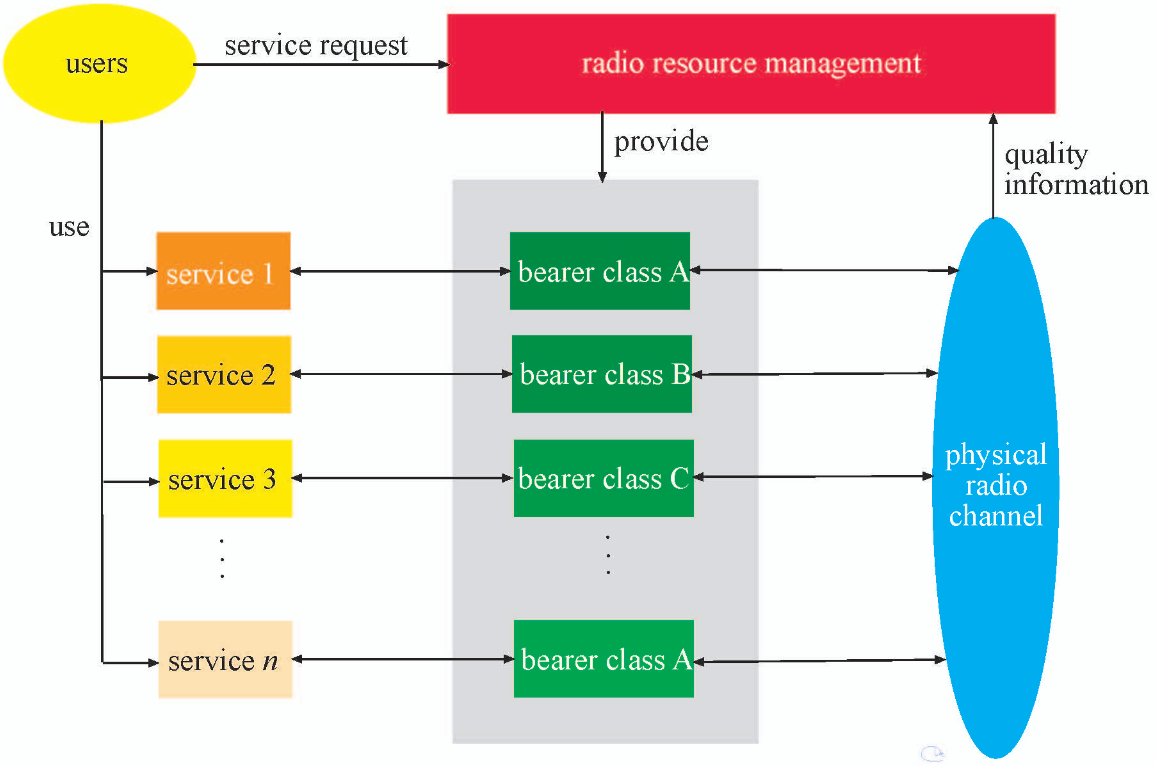


*Figure 2.2 Generic radio resource allocation for ASD*

## 2.3 Adaptive service deployment (ASD) approach

To facilitate an improved joint spectral efficiency, an adaptive service deployment (ASD) approach can be used. The ASD approach uses a hybrid channel allocation (HCA) scheme which overcomes the weaknesses of fixed channel allocation (FCA), widely deployed in second generation cellular systems, and of dynamic channel allocation (DCA), which is used in e.g. the Digital Enhanced Cordless Telecommunications (DECT) system. The FCA part of the HCA scheme uses a classical frequency planning similar to the one addressed in Sect. 2.2. Simulations have shown that a cluster order $r$ of three provides a very stable radio network with a high spectral efficiency for most service cases. Therefore, cluster order $r$ equal to three is proposed for WB-TDMA/CDMA.

In order to enable the network to deploy the available resources as efficiently as possible, the DCA part of the aforementioned HCA scheme is based on link quality measurements. This will be outlined in what follows. The choice of the parameters of a bearer class, in particular of the link level parameters shown in *Figure 2.1*, is based on a variety of measurement results of so-called quality parameters, such as measured distance between mobiles and base transceiver stations, measured signal-to-noise ratio or carrier-to-interference ratio at the detector inputs, measured duration of the channel impulse response, estimated bit error ratio, and measured mobile speeds. Setting out from these measurement results, the base station controller allocates the radio resources for each of the requested services and for each mobile user in an adaptive fashion. This approach facilitates the on-the-fly maximization of the spectral efficiency.

*Figure 2.2* illustrates the radio resource allocation for WB-TDMA/CDMA in a generic way. The users wish to access a variety of services 1 to $n$. Therefore, random access requests are transmitted to the base transceive stations which direct these requests to the base station controller. In addition, signaling between base transceive stations and mobiles facilitates the measurement of the aforementioned quality parameters. This quality information is directly related to the physical radio channel. Based on the requests for service utilization by the users and the available quality information, the base station controllers decide on the granting of the requested accesses based on the availability of the services in both the network and the environment. Furthermore, the parameters of the bearer classes are negotiated.

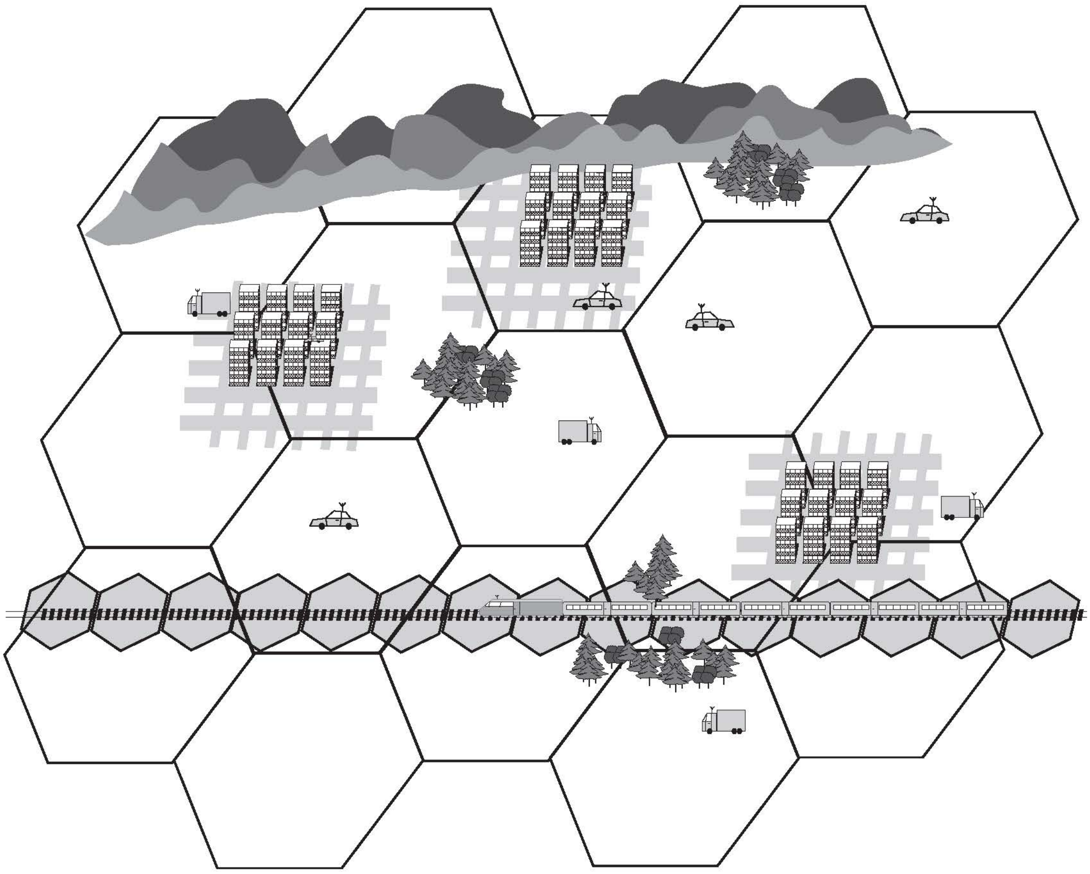

*Figure 2.3 Part of a hierarchical cell structures (HCS) environment*

The ASD approach can be further extended when considering hierarchical cell structures (HCS). Different services classes which are associated with different bearer classes can be provided in different cell layers. This allows for an additional means of optimization of the network structure. By setting out from the measurement results of the quality parameters addressed before, the allocation of users to the different hierarchies in the HCS environment is done. The HCA based resource allocation will then be done within each cell layer.

In *Figure 2.3*, a part of a HCS environment is depicted. The lowest hierarchy is presented by a network of macro cells which are assumed to be hexagonal. Within the macro cells, Manhattan grid like micro cellular structures are located. Within buildings, pico cells can be used. Furthermore, special small macro cells are used to provide service to e.g. high speed trains or motorways. The ASD approach supports seamless handover between different cell layers and within the cell layers. The handover is based on the measurement results of the aforementioned quality parameters.

The ASD approach allows the utilization of different services with varying QoS criteria by various users when simultaneously maintaining a high spectral efficiency. Also, different services with different QoS criteria can be used by a particular user with high spectral efficiency.

# 3. Examples of mapping of bearer services on physical channels

## *3.1 Overview*

In the case of WB-TDMA/CDMA, the total number of available basic physical channels per TDMA frame is given by the maximum number of time slots which is 8 and the maximum number of CDMA codes per time slot which is 8 in case the different codes within one time slot are allocated to different users in the uplink and which is 12 in the downlink and 9 in case the different codes within one time slot are allocated to one and the same user in the up- and downlink. The total number of basic physical