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Abstract of document:

This document is a technical report titled 'Feasibility Study for Enhanced Uplink for UTRA FDD' for the Release 6 study item “Uplink Enhancements for Dedicated Transport Channels"

Changes since last presentation to TSG RAN:

This is the first presentation of this document to TSG RAN

Outstanding Issues:

Clarification of outstanding issues are presented in [1]

Contentious Issues:

Clarification of contentious issues are presented in [1]

References:

[1] RP-030399, Status Report for SI on Uplink Enhancements for Dedicated Transport Channels

3GPP TR 25.896 V1.0.0 (2003-09)

Technical Report

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Contents

Foreword

This Technical Report has been produced by the 3rd Generation Partnership Project (3GPP).

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y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the document.

1 Scope

This present document is the technical report for the Release 6 study item "Uplink Enhancements for Dedicated Transport Channels"(see [1]).

The purpose of this TR is to help TSG RAN WG1 to define and describe the potential enhancements under consideration and compare the benefits of each enhancement with earlier releases for improving the performance of the dedicated transport channels in UTRA FDD uplink, along with the complexity evaluation of each technique. The scope is to either enhance uplink performance in general or to enhance the uplink performance for background, interactive and streaming based traffic.

This activity involves the Radio Access work area of the 3GPP studies and has impacts both on the Mobile Equipment and Access Network of the 3GPP systems.

This document is intended to gather all information in order to compare the solutions and gains vs. complexity, and draw a conclusion on way forward.

This document is a 'living' document, i.e. it is permanently updated and presented to TSG-RAN meetings.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TD RP-020658: "Study Item Description for Uplink Enhancements for Dedicated Transport Channels ".

[2] 3GPP RAN WG1 TDOC R1-00-0909, "Evaluation Methods for High Speed Downlink Packet Access (HSDPA)", July 4 2000

[3] Hämäläinen S., P. Slanina, M. Hartman, A. Lappeteläinen, H. Holma, O. Salonaho, "A Novel Interface between Link and System Level Simulations", Proceedings of ACTS summit 1997, Aalborg, Denmark, Oct. 1997, pp. 509-604.

[4] 3GPP RAN WG1#29 TDOC R1-02-1326, "Link Prediction methodology for System Level Simulations", Shanghai China, November 5 2002.

[5] Ratasuk, Ghosh, Classon, "Quasi-Static Method for Predicting Link-Level Performance" IEEE VTC 2002.

[6] 3GPP TR 25.942 V3.3.0 (2002-06), RF System Scenarios, June 2002.

[7] 3GPP TR 25.853 V1.3.0 (2003-03), "Delay Budget within the Access Stratum", March 2003.

[8] 3GPP TS 25.133 V3.11.0 (2002-09), "Requirements for support of radio resource management (FDD) (Release 99)", September 2002.

[9] Hytönen, T.; "Optimal Wrap-around Network Simulation", Helsinki University of Technology Institute of Mathematics Research Reports, 2001, www.math.hut.fi/reports/, Report number A432

[10] "Source Models of Network Game Traffic", M. S. Borella, Proceedings, Networld+Interop '99 Engineer's Conference, May 1999.

[11] 3GPP RAN WG1#30 TDOC R1-03-0083, "Link Prediction Methodology for System Level Simlations," Lucent Technologies, San Diego, USA, January 7-10, 2003.

[12] 3GPP2, 1xEV-DV Evaluation Methodology.

[13] ETSI TR 101 12, Universal Mobile Telecommunications System (UMTS); Selection procedures for the choice of radio transmission technologies of the UMTS (UMTS 30.03 v3.2.0)

[14] TS 25.214, v5.3.0, "Physical layer procedures (FDD)", December 2002

[15] TS 25.331, v5.4.0, "Radio Resource Control (RRC); Protocol Specification", March 2003

3 Definitions, symbols and abbreviations

E-DCH Enhanced DCH, a new dedicated transport channel type or enhancements to an existing dedicated transport channel type (if required by a particular proposal)

E-DPCCH Enhanced DPCCH, a physical control channel associated with the E-DPDCH (if required by a particular proposal)

E-DPDCH Enhanced DPDCH, a new physical data channel or enhancements to the current DPDCH (if required by a particular proposal)

4 Introduction

At the 3GPP TSG RAN #17 meeting, SI description on "Uplink Enhancements for Dedicated Transport Channels" was approved [1].

The justification of the study item was, that since the use of IP based services becomes more important there is an increasing demand to improve the coverage and throughput as well as reduce the delay of the uplink. Applications that could benefit from an enhanced uplink may include services like video-clips, multimedia, e-mail, telematics, gaming, video-streaming etc. This study item investigates enhancements that can be applied to UTRA in order to improve the performance on uplink dedicated transport channels.

The study includes, but is not restricted to, the following topics related to enhanced uplink for UTRA FDD to enhance uplink performance in general or to enhance the uplink performance for background, interactive and streaming based traffic:

- Adaptive modulation and coding schemes
- Hybrid ARQ protocols
- Node B controlled scheduling
- Physical layer or higher layer signalling mechanisms to support the enhancements
- Fast DCH setup
- Shorter frame size and improved QoS

5 Requirements

- The overall goal is to improve the coverage and throughput as well as to reduce the delay of the uplink dedicated transport channels.
- The focus shall be on urban, sub-urban and rural deployment scenarios. Full mobility shall be supported, i.e., mobility should be supported for high-speed cases also, but optimisation should be for low-speed to medium-speed scenarios.
- The study shall investigate the possibilities to enhance the uplink performance on the dedicated transport channels in general, with priority to streaming, interactive and background services.
- Features or group of features should demonstrate significant incremental gain, with reasonable complexity. The value added per feature should be considered in the evaluation.
- The UE and network complexity shall be minimised for a given level of system performance.
- The impact on current releases in terms of both protocol and hardware perspectives shall be taken into account.
- It shall be possible to introduce the new features in the network which has terminals from Release'99, Release 4 or Release 5.

6 Reference Techniques in Earlier 3GPP Releases

Editor's Note: This chapter shall contain the description of current techniques specified in earlier 3GPP standard releases for background information and for reference to compare proposed new techniques to. The reference techniques should contain e.g. current scheduling techniques and DCH setup mechanisms.

6.1 DCH Setup Mechanisms

A fundamental concept in WCDMA is the connection state model, illustrated in Figure 6.1.1. The connection state model enables optimization of radio and hardware resources depending on the activity level of each UE.

- Users with high transmission activity (in either uplink, downlink or both) should be in CELL_DCH state, where power-controlled dedicated channels are established to/from the UE. In CELL_DCH state, the UE is assigned dedicated radio and hardware resources, which minimizes processing delay and allows for high capacity.

- Users with low transmission activity should be in CELL_FACH state, where only common channels are used. The major advantages with CELL_FACH state are the possibility for low UE power consumption and that no dedicated hardware resources in the Node B are needed.

- Users with no transmission activity are in CELL_PCH or URA_PCH states, which enable very low UE power consumption but do not allow any data transmission. These states are not discussed further in this section.

Switching between CELL_DCH and CELL_FACH are controlled by the RRC based on requests from either the network or the UE. Entering CELL_DCH implies the establishment of a DCH, which involves a physical layer random access procedure, NBAP and RRC signaling, and uplink and downlink physical channel synchronization.

Clearly, it is desirable to switch a UE to CELL_FACH state when there is little transmission activity in order to save network resources and to reduce the UE power consumption. Switching between CELL_DCH and CELL_FACH is especially useful in scenarios with a large number of bursty packet data users, where there is a risk that the system becomes code limited if users temporarily not receiving/transmitting any packets are not switched to CELL_FACH. When the activity increases, the UE should rapidly be switched back to CELL_DCH and a dedicated channel be established.

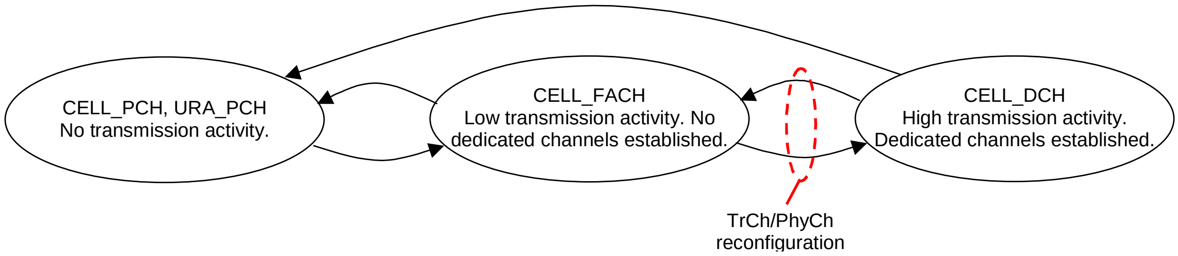

Figure 6.1.1: Connection states.

6.1.1 Uplink/Downlink Synchronization

The DCH setup procedure in Rel99/4/5 is illustrated in Figure 6.1.2. At time t_1, downlink data arrives to the RNC and a decision to establish a DCH is taken at time t_2. The decision is sent to the UE via the S-CCPCH, which starts to establish synchronization to the downlink DPCCH at time t_4, using the standardized procedure described in [14].

The downlink synchronization procedure is divided into two phases: The first phase starts when higher layers in the UE initiate physical dedicated channel establishment and lasts until 160 ms after the downlink dedicated channel is considered established by higher layers. During this time, out-of-sync shall not be reported and in-sync shall be reported using the CPHY-Sync-IND primitive if the downlink DPCCH quality exceeds a threshold for at least 40 ms. The second phase starts 160 ms after the downlink dedicated channel is considered established by higher layers. During this phase, both out-of-sync and in-sync are reported, depending on the situation in the UE. As the UE is not allowed to report in-sync until at least 40 ms after the start of the first synchronization phase, the interval T_4 equals at least 40 ms.

Once the UE has detected the in-sync condition for the downlink DPCCH, the UE starts transmitting the uplink power control preamble at time t_5. The length of the power control preamble, T_5, is set by higher layer signaling. During this period, the Node B establishes synchronization with the UE on the uplink. Once the power control preamble is finished, at t_6, the UE uplink/downlink DPCH is established and data transmission may begin.

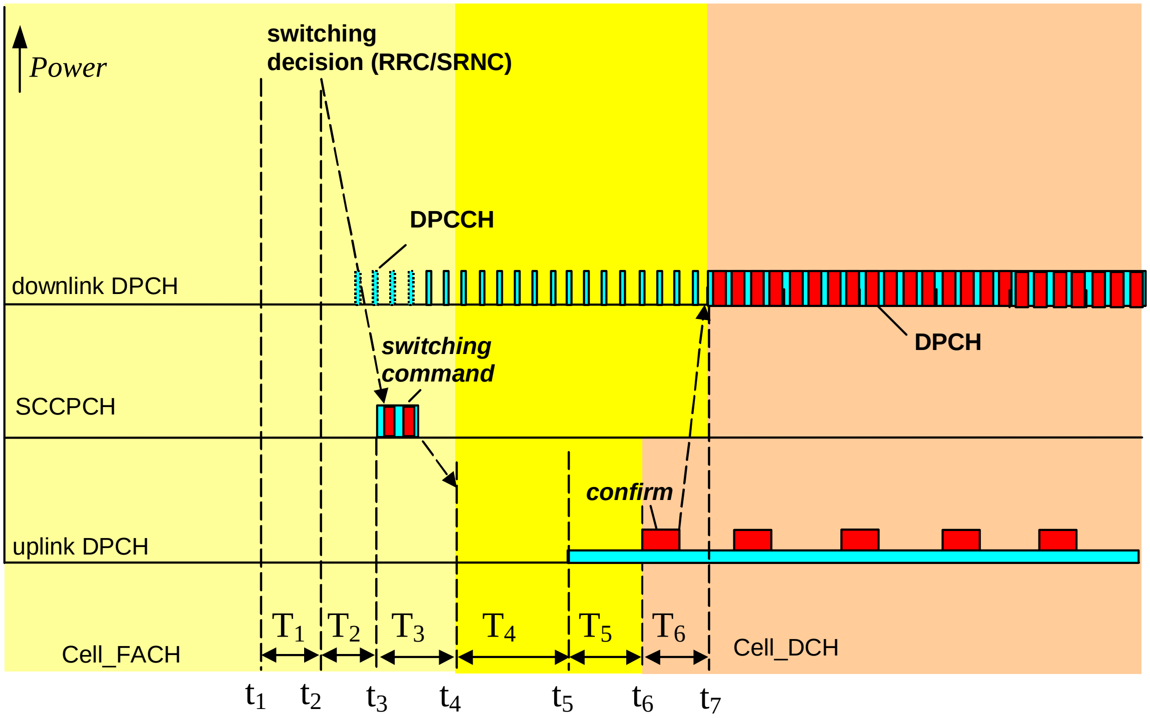

Figure 6.1.2: Rel99/4/5 procedure for DCH establishment. Note that T_6 is optional and data transmission may start already at t_6.

6.2 Uplink TFCS Management with RRC Signalling

There are following TFCS reconfiguration messages available in current specifications [1]:

- Complete reconfiguration, in which case UE shall remove a previously stored TFCS set, if it exists
- Addition, in which case UE shall insert the new additional TFC(s) into the first available position(s) in ascending order in the TFCS.
- Removal, in which case UE shall remove the TFC indicated by “IE” TFCI from the current TFCS, and regard this position (TFCI) as vacant.
- Replace, in which case UE shall replace the TFCs indicated by “IE” TFCI and replace them with the defined new TFCs.

In addition to those, there is also Transport format combination control message defined in [1], with which the network can define certain restrictions in the earlier defined TFCS set, as described below.

- Transport Format Combination Subset in the TFC control message can be defined in the format of TFCS restriction; for downgrading the original TFCS set. There are several different formats possible. The message can define the minimum allowed TFC index in the original TFCS set. Or it can define that a certain TFC subset from the original TFCS set is either allowed or not. One possible way to define the message is to list what Transport channels have restrictions, and then list the allowed TFIs for the restricted Transport channels.
- Transport Format Combination Subset in the TFC control message message can be defined in the format of canceling the earlier TFCS restriction; i.e. defining that the original TFCS set is valid again.

Transport format combination control message includes activation time. The activation time defines the frame number /time at which the changes caused by the related message shall take effect. The activation time can be defined as a function of CFN, ranging between 0…255, the default being “now”.

Transport format combination control message can also include an optional parameter of TFC control duration, which defines the period in multiples of 10 ms frames for which the defined restriction, i.e. TFC subset , is to be applied. The possible values for this are (1,2,4,8,16,24,32,48,64,128,192,256,512).

In [1], in chapter 13.5, it is defined separately for each RRC procedure, what kind of delay requirements there are for UE. For TFCS control messages there are following delay requirements:

- TRANSPORT FORMAT COMBINATION CONTROL : N1 = 5 . This defines the upper limit on the time required to execute modifications in UE after the reception of the RRC message has been completed. This means that after receiving the TFCS control message, the UE shall adopt the changes in the beginning of the next TTI starting after N1*10ms .
- TRANSPORT FORMAT COMBINATION CONTROL FAILURE: N2=8. This defines the number of 10 ms radio frames from end of reception of UTRAN -> UE message on UE physical layer before the transmission of the UE -> UTRAN response message must be ready to start on a transport channel with no access delay other than the TTI alignment. The UE response message transmission from the physical layer shall begin at the latest (N2*10)+TTI ms after completion of the reception of the last TTI carrying the triggering UTRAN -> UE message. When Target State is CELL_DCH, the UE response message transmission from the physical layer may be additionally delayed by the value of IE "SRB delay".

6.3 Transport Format Combination Selection in the UE

6.3.1 Description of TFC selection method

The TFC selection is a MAC function that the UE uses to select a TFC from its current TFCS whenever it has something to transmit. The TFC is selected based on the need for data rate (i.e. UE buffer contents), the currently available transmission power, the available TFCS and the UE's capabilities. The details of the TFC selection function are covered in [2] and [3].

The most important parameters governing the behavior of the TFC selection function are called X,Y and Z, and their values have been agreed to be static in the current specifications. Table 6.3.1 below shows the values of these parameters.

Table 6.3.1: X, Y, Z parameters for TFC selection

X	Y	Z
15	30	30

Based on these parameters, the UE shall continuously evaluate based on the *Elimination*, *Recovery* and *Blocking* criteria defined below, how TFCs on an uplink DPDCH can be used for the purpose of TFC selection. The following diagram illustrates the state transitions for the state of a given TFC.

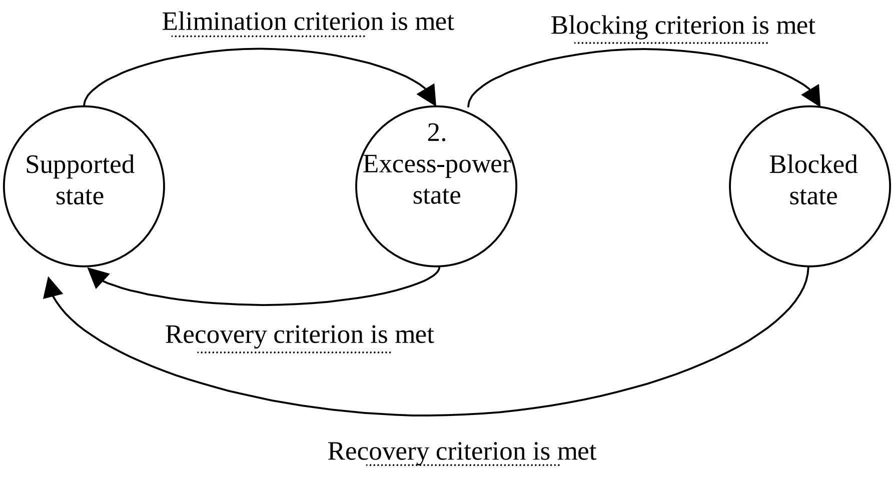

Figure 6.3.1: State transitions for the state of a given TFC

The evaluation shall be performed for every TFC in the TFCS using the estimated UE transmit power. The UE transmit power estimation for a given TFC shall be made using the UE transmitted power measured over the measurement period, defined in section 9.1.6.1 of [2] as one slot, and the gain factors of the corresponding TFC. Table 6.3.2 below, extracted from [2], shows the specified accuracy requirements for measuring UE transmit power over the one slot measurement period, as a function of the current transmit power level relative to maximum output power.

Table 6.3.2: UE transmitted power absolute accuracy

Parameter	Unit	Accuracy [dB]	
		PUEMAX 24dBm	PUEMAX 21dBm
UE transmitted power=PUEMAX	dBm	+1/-3	±2
UE transmitted power=PUEMAX-1	dBm	+1.5/-3.5	±2.5
UE transmitted power=PUEMAX-2	dBm	+2/-4	±3
UE transmitted power=PUEMAX-3	dBm	+2.5/-4.5	±3.5
PUEMAX-10≤UE transmitted power<PUEMAX-3	dBm	+3/-5	±4

NOTE 1: User equipment maximum output power, PUEMAX, is the maximum output power level without tolerance defined for the power class of the UE in TS 25.101, section 6.2.1.

The UE shall consider the *Elimination* criterion for a given TFC to be detected if the estimated UE transmit power needed for this TFC is greater than the Maximum UE transmitter power for at least X out of the last Y successive measurement periods immediately preceding evaluation. The MAC in the UE shall consider that the TFC is in Excess-Power state for the purpose of TFC selection.

MAC in the UE shall indicate the available bitrate for each logical channel to upper layers within T_{notify} from the moment the *Elimination* criterion was detected.

The UE shall consider the *Recovery* criterion for a given TFC to be detected if the estimated UE transmit power needed for this TFC has not been greater than the Maximum UE transmitter power for the last Z successive measurement periods immediately preceding evaluation. The MAC in the UE shall consider that the TFC is in Supported state for the purpose of TFC selection.

MAC in the UE shall indicate the available bitrate for each logical channel to upper layers within T_{notify} from the moment the *Recovery* criterion was detected.

The evaluation of the *Elimination* criterion and the *Recovery* criterion shall be performed at least once per radio frame.

The UE shall consider the *Blocking* criterion for a given TFC to be fulfilled at the latest at the start of the longest uplink TTI after the moment at which the TFC will have been in Excess-Power state for a duration of:

$(T_{notify} + T_{modify} + T_{L1\_proc})$

where:

T_{notify} equals 15 ms

T_{modify} equals $MAX(T_{adapt\_max}, T_{TTI})$

$T_{L1\ proc}$ equals 15 ms

T_{adapt_max} equals $MAX(T_{adapt\_1}, T_{adapt\_2}, ..., T_{adapt\_N})$

N equals the number of logical channels that need to change rate

T_{adapt_n} equals the time it takes for higher layers to provide data to MAC in a new supported bitrate,

for logical channel n. Table 6.3.3 defines T_{adapt} times for different services. For services where no codec is used T_{adapt} shall be considered to be equal to 0 ms.

Table 6.3.3: T_{adapt}

Service	T_{adapt} [ms]
UMTS AMR	60
UMTS AMR2	60

T_{TTI} equals the longest uplink TTI of the selected TFC (ms).

Before selecting a TFC, i.e. at every boundary of the shortest TTI, the set of valid TFCs shall be established. All TFCs in the set of valid TFCs shall:

1. belong to the TFCS.
2. not be in the Blocked state.
3. be compatible with the RLC configuration.
4. not require RLC to produce padding PDUs
5. not carry more bits than can be transmitted in a TTI (e.g. when compressed mode by higher layer scheduling is used and the presence of compressed frames reduces the number of bits that can be transmitted in a TTI using the Minimum SF configured).

The UE may remove from the set of valid TFCs, TFCs in Excess-power state in order to maintain the quality of service for sensitive applications (e.g. speech). Additionally, if compressed frames are present within the longest configured TTI to which the next transmission belongs, the UE may remove TFCs from the set of valid TFCs in order to account for the higher power requirements.

The chosen TFC shall be selected from within the set of valid TFCs and shall satisfy the following criteria in the order in which they are listed below:

1. No other TFC shall allow the transmission of more highest priority data than the chosen TFC.
2. No other TFC shall allow the transmission of more data from the next lower priority logical channels. Apply this criterion recursively for the remaining priority levels.
3. No other TFC shall have a lower bit rate than the chosen TFC.

The above rules for TFC selection in the UE shall apply to DCH, and the same rules shall apply for TF selection on RACH and CPCH.

UE shall consider that the Blocking criterion is never met for TFCs included in the minimum set of TFCs (see [4]).

6.3.2 TFC selection method as a reference case for Enhanced Uplink DCH

The important parameters to be included to the simulation assumptions for TFC selection method in the reference case are:

a) Accuracy of the UE transmit power estimate. See table 6.3.2 in the previous section as a reference. This will have an effect how fast UE moves a certain TFC to excess power state. Since the accuracy depends on the currently used transmit power level, it is noted for the purpose of general understanding, that the accuracy is thus in average worse with a bursty traffic model, in which quite often only DPCCH is transmitted, than with more real-time type of application in which transmission of DPDCH is more continuous. Also the location in the cell will effect to the accuracy due to the same reason. It is however seen that for the sake of simplicity, it would be appropriate to define only one value for this parameter used in all simulations.

   It is thus proposed that the accuracy defined for the maximum Ptx power level, ±2 dB, is used in all cases, for the sake of simplicity of the simulations. This is to be modelled so that the error is lognormally distributed with zero mean and std=1.2159 dB, which has the effect of causing 90% of the errors to occur within ±2 dB of the zero mean. It is noted that the accuracy requirements in [2] are also defined for 90% probability.

b) Delay between the moment when elimination criterion is met in L1 and when the TFC is moved into blocked state. See the previous section as a reference, together with the Annex A.6.4.2.1 from [2], defining the maximum delay to be $T_{notify} + T_{modify} + T_{L1\_proc} + T_{align\_TTI}$. In addition to this , if criterion is met with a maximum misalignment between the frame boundary, an extra 14 slots (9.33 ms) will need to be added to this delay. It is proposed that in the simulation assumptions the assumption is that there is no codec (e.g. AMR) involved, the rate of which should be adjusted and that the longest TTI in the selected TFC is T_{TTI} =10 ms= T_{modify}. This will result in a maximum delay of (9.33 ms + $T_{notify} + T_{modify} + T_{L1\_proc} + T_{align\_TTI}$) = (9.33 + 15 + 10 + 15 + 10) ms= 59.33 ms.

c) Delay between the moment recovery criterion is met and when TFC is moved back to supported state. See the previous section as a reference, together with the Annex A.6.4.2.1 from [2], defining the maximum delay to be $T_{notify} + T_{modify} + T_{L1\_proc} + T_{align\_TTI.}$ In addition to this , if criterion is met with a maximum misalignment between the frame boundary, an extra 14 slots (9.33 ms) will need to be added to this delay. It is proposed that in the simulation assumptions the assumption is that there is no codec (e.g. AMR) involved, the rate of which should be adjusted and that the longest TTI in the selected TFC is T_{TTI} =10 ms= T_{modify}. This will result in a maximum delay of (9.33 ms + $T_{notify} + T_{modify} + T_{L1\_proc} + T_{align\_TTI}$) = (9.33 + 15 + 10 + 15 + 10) ms= 59.33 ms.

d) TFCS ; i.e. the set of allowed user bit rates allocated to the UE. These are the bit rates that UE can use in the TFC selection algorithm. There should be enough steps in the TFCS to allow the UE to decrease the used data rate in a flexible fashion at the cell edge. It is proposed that there are two TFCS sets used in the reference case: [8, 16, 32, 64, 128, 256, 384] kbit/s and [8, 16, 32, 64, 128, 256, 384, 768, 1000] kbit/s . The idea why to have 2 sets is to allow to study different peak data rate in the proposed schemes with a sensible TFCS set in the reference case to be compared with.

The parameters and parameter values explained above are inserted to the Annex A.3, System simulation assumptions, Table A - 8 - System Level Simulation parameters used in the reference rel99/rel4/rel5 case.

It is noted that TFC selection method should be modelled also in the new schemes proposed for Enhanced Uplink DCH, if there is no clear reason why it can not/should not be included into the proposed scheme. The parameters used should be the same, or at least similar (e.g. TFCS set), as defined in the reference case.

7 Overview of Techniques considered to support Enhanced Uplink

7.1 Scheduling <NodeB controlled scheduling, AMC>

Two fundamental approaches exist to scheduling UE transmissions for the E-DCH – pure rate scheduling, where all uplink transmissions occur in parallel, but at a low enough rate that the desired total noise rise at the NodeB is not exceeded, and pure time scheduling, where theoretically only a subset of UEs that has traffic to send is allowed to transmit at any given time, again such that the desired total noise rise at the Node B is not exceeded. For rate scheduling, restricting the rate to control the noise rise in effect restricts the UE transmit power. For time scheduling, the UEs may be selected on the basis of uplink channel conditions.

The usage of either rate or time scheduling is of course restricted by available power because the E-DCH will have to co-exist with a mix of other transmissions by that UE and other UEs in the uplink. A hybrid of these two approaches is of course also possible, where different proposals will tend to favour one or other of the fundamental approaches.

7.1.1 Node B Controlled Rate Scheduling by Fast TFCS Restriction Control

Editor's Note: This chapter is currently describing one possible solution for Node B controlled scheduling using a new L1 mechanism for transport format combination control. Other possible solutions may be defined later.

7.1.1.1 Purpose and General Assumptions

The purpose of the studied technique is to enable more efficient use of the uplink power resource of the cell in order to provide a higher cell throughput in the uplink and a larger coverage area for higher uplink data rates for streaming, interactive and background class services. These goals are to be reached by fast Node B controlled uplink scheduling which provides a better control to uplink noise rise and enables better control to noise rise variance.

In the existing Rel'99/Rel'4/Rel'5 system the uplink scheduling and data rate control resides in the RNC, which is not able to respond to the changes in the uplink load as fast as a control residing in Node B could. Thus the Node B control is seen to be requiring less UL noise rise headroom for combatting overload conditions. Node B control is also seen capable of smoothing the noise rise variance by allocating higher data rates quickly when the uplink load decreases and respectively by restricting the uplink data rates when the uplink load increases.

This enhancement technique is a method which itself does not require changes to the uplink DCH structure but rather introduces new L1 signalling to facilitate fast UL scheduling by means of transport format combination control. Hence the method does not require a new transport channel to be defined, but does not forbid it either. The method can be applied with or without other enhancements such as for example HARQ and Fast DCH Setup.

7.1.1.2 General Principle

The basic principle of the technique is to allow Node B set and control a new restriction to the TFC selection mechanism of the UE by fast L1 signalling. From the UE point of view the scheduling principle is the same than in existing Rel'99/Rel'4/Rel'5 system with the modification that there would be additional L1 control over a new restriction to its TFC selection mechanism. In the UTRAN side, a new scheduling by the means of fast TFCS restriction control is introduced in Node B.

All the same functions considered for the enhancement technique can be achieved with already existing RRC procedures for TFCS configuration and transport format combination control. However, by allowing the Node B to have control over TFCS restrictions (i.e. provide a mechanims for transport format combination contorol in L1) enhances the speed of which the UTRA can adapt to the changes in the UL load. In Rel'99/Rel'4/Rel'5, restricting the set of alowed TFCs in a TFCS is done using an RRC signalling procedure called transport format combination control.

7.1.1.3 Restricting the Allowed Uplink TFCs in a TFCS by L1 Signalling

In the subsequent chapters, a new mechanism and related L1 signalling are introduced. The purpose is to enable the Node B to have a fast control over the TFC subset allowed to be used by the TFC selection algorithm of the UE. This is to be achieved by defining two TFC subsets of the TFCS (A "Node B allowed TFC subset" and a "UE allowed TFC subset"), and control signalling for adjusting these subsets.

Node B provides UE with an allowed TFC subset" from which the UE's TFC selection algorithm selects a TFC to be used by employing the TFC selection method defined in Rel'99/Rel'4/Rel'5 specifications. This TFC subset provided by the Node B is is named the “UE allowed TFC subset”.

In order to give RNC efficient control over the "UE allowed TFC subset" primarily controlled by the Node B, the RNC provides the Node B with a second TFC subset named “Node B allowed TFC subset”. Node B defines and freely reconfigures the "UE allowed TFC subset" as a subset of the "Node B allowed TFC subset". It is expected that with the “Node B allowed TFC subset” RNC is able to do similar TFC restrictions as done in Rel'99/Rel'4/Rel'5 by using Transport Format Combination Control procedure defined in RRC signalling. Both subsets are defined individually for each UE.

The “UE allowed TFC subset” and the “Node B allowed TFC subset” may be signalled in the form of TFC pointers pointing to the TFCS of the UE, if the TFCs can be arranged in an order that corresponds to the TFC restriction rule (or scheduling strategy) that the Node B would be willing to apply. The ordering rule may be explicit or implicit.

In a example illustrated in the Figure7.1.1 below the Node B may want to restrict the TFCs is the order of Tx power for the CCTrCH. In Figure 7.1.1, the TFCs in a TFCS are shown ordered in descending order (with respect to the power required) starting from zero. Both TFC pointers are initialised to both the Node B and to the UE by the RNC in the beginning of the connection. After initialisation the Node B can command the UE pointer up/down with the restriction that UE pointer may not exceed Node B pointer. The TFC selection algorithm in the UE may select any TFC up to the TFC indicated by the UE pointer. The purpose here is to control the UE's power usage by restricting it's TFC (i.e. data rate) selection.

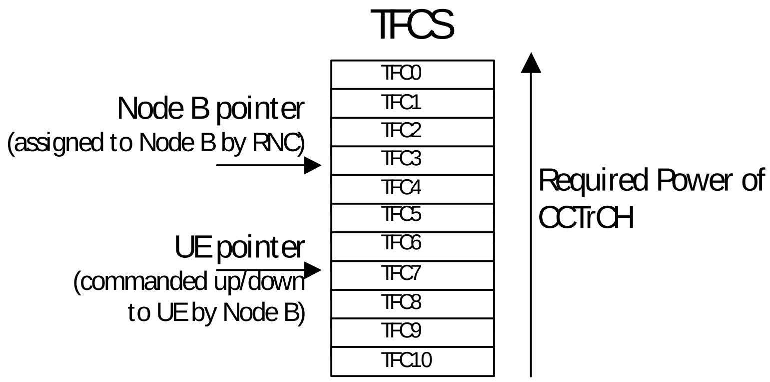

Figure 7.1.1: Depiction of the TFC pointers

The UE and Node B allowed TFC subsets should not restrict the use of the TFCs in the minimum TFC set guaranteed to be available for UE's TFC selection at all times unless the minimum TFC set definition in the already existing specifications is changed. (Minimum TFC set is defined in Rel'99/Rel'4/Rel'5 specifications)

7.1.1.4 Issues Requiring Further Studying

It is FFS, how a DCH controlled with this method could be multiplexed with DCHs controlled with Rel'99/Rel'4/Rel'5 methods, especially keeping in mind that simultaneous conversational traffic should be possible. Methods for using separate code channel and TFCS, as well as multiplexing the Node B controlled DCH with e.g. a DCH carrying voice in the same CCTrCH are to be studied. Naturally, if a DCH carrying e.g. conversational traffic is multiplexed with a DCH carrying streaming, interactive or background traffic, the first DCH carrying conversational traffic still represents the non-controllable load and only the second DCH could be controlled by the proposed method.

It is FFS how the method should work in different reconfiguration cases, such as physical channel and transport channel reconfigurations, TFCS reconfiguration for the UE, Node B allowed TFC subset reconfiguration for the Node B. E.g. in TFCS reconfiguration it should be defined whether UE continues the transmission with the new "UE allowed TFC subset", or continues with the old one. To allow flexible update of "Node B allowed TFC subset" to the Node B, and simultaneously minimise the amount of RRC signaling, one possibility is that "Node B allowed TFC subset" is not informed to the UE at all.

It is also FFS how the method should work in soft handover. One possibility in the event the use of two pointers is applicable is to use the same kind of method as defined for TPC commands. I.e. each cell in the active set receives L1 signalling from the UE and transmits L1 signalling to the UE independently from the other cells. Only if all the cells in the active set command the UE pointer increment, the UE increases the UE pointer with one step. Respectively, if at least one Node B in the active set commands the UE pointer decrement, the UE decreases the UE pointer (and therefore the maximum power that can be transmitted) with one step. Also other possibilities exist and should be investigated.

The impacts of L1 signalling errors (including possible error accumulation) is FFS. This includes possible mitigation techniques. Both the non-SHO and the SHO cases need to be considered.

7.1.1.5 Signalling to Support Fast TFCS Restriction Control

Editor's Note: This chapter is currently describing one possible solution for the signalling to support the method. Other possible signalling solutions may be introduced later.

7.1.1.5.1 L1 signaling

Two new L1 messages are introduced in order to enable the transport format combination control by L1 signalling between the Node B and the UE.

- Rate Request (RR), sent in the uplink by the UE to the Node B. With the RR the UE can ask the Node B to change the set of the allowed uplink transport format combinations within the transport format combination set.

- Rate Grant (RG), sent in the downlink by the Node B to the UE. With RG, the Node B can change the allowed uplink transport format combinations within the transport format combination set.

7.1.1.5.2 RRC signalling

Editor's Note: This chapter is to be defined later.

7.1.1.5.3 Iub/Iur signalling

Editor's Note: This chapter is to be defined later.

7.1.2 Method for Node B Controlled Time and Rate Scheduling

7.1.2.1 Purpose and General Assumptions

Current UMTS R99/R4/R5 DCH specifications support autonomous UE transmission and UE TFCS control using Radio Resource Control (RRC) messaging to establish and manage a per UE Transport Format Combination Set (TFCS). TFCS reconfiguration latency and update rate is restricted by the communication delay between the RNC and Node-B since the TFCS reconfiguration function is centralized in the RNC. Besides using more frequent and lower latency TFCS updates to better manage uplink interference, additional advantages are possible by controlling the time at which UEs transmit compared to allowing autonomous UE transmissions. If TFCS control is to be shared between the RNC and Node B to enable fast TFCS control and higher UE uplink data rates are to be supported, then controlling time of UE transmissions may also be necessary to most efficiently and correctly control uplink intereference levels for maximizing throughput.

7.1.2.2 General Principle

The basic principle of the technique is to allow Node B control of UE TFCS and UE transmission time by fast L1 signalling. The difference to existing R99/R4/R5 systems is that the UE would receive additional L1 control over its TFC selection and L1 control of its transmission time. From the UTRAN’s perspective, scheduling by means of TFCS indicator and transmission time control is introduced at the Node B. A UE is sent a scheduling assignment by a scheduling Node B. The UE transmits during the time interval specified by the downlink scheduling assignment using a restricted TFCS, which is determined from a TFCS indicator in the scheduling assignment. It is possible to make use of existing RRC procedures for TFCS configuration and transport format combination control and utilize them at the Node B for determining a TFC. RNC and Node B control of UE TFCS and transmission time allows the UTRAN to control the changes in the UL load.

7.1.2.3 Controlling UE TFCS and transmission time

In the subsequent chapters, a new mechanism for scheduling and related L1 signalling is introduced. The purpose is to enable the Node-B to explicitly determine when and which UE’s should transmit data on the uplink and to control the TFCS at each scheduled UE to control the uplink interference level and variation.

Instead of a Node-B continously controlling each UE’s TFCS by sending up/down adjustments to a pointer, the Node-B sends a TFCS indicator (which could be a pointer e.g.) in the signaled scheduling assignment. The scheduling assignment also indicates the scheduling time interval over which the UE must transmit given it has non-zero buffer occupancy. The TFCS indicator specifies the TFC(s) corresponding to the highest rate/power level the UE is allowed to transmit at during the specified time interval. After the scheduled time interval has elapsed, the TFCS reverts back to the set that existed prior to the scheduled time interval. A scheduled UE is allowed to choose among the TFCs in the restricted TFCS in terms of rate and power as determined by the TFCS indicator and based upon its own status e.g. actual available power and latest buffer status. The rates used by the UE could be signaled on the associated uplink signalling channel e.g. E-DPCCH at the time of transmission. Uplink power control information received by each UE may be used to effectively adjust the TFCS indicator over the scheduling interval.

The Node B may decide which UE(s) are allowed to transmit and the corresponding TFCS indicators on a per TTI basis based on, for example, some knowledge of the following:

- Buffer status of each UE

- Power status of each UE[1]
- Local Node B measured channel quality estimate for each UE[2] or maximum UE power capability at Node B.
- Available interference Rise Over Thermal (RoT) margin (or threshold level) at the Node B

The RoT margin may be computed by taking into account the thermal noise, other cell interference (Ioc), the Eb/No requirements for power controlled (e.g. voice) channels (see Figure 7.1.2) and information provided by the RNC.

Node B Controlled Time and Rate scheduling may have several advantages. Reduced latencies in rate control, exploitation of fast channel quality variations, more precise RoT control (i.e., better interference management), and consequently, better efficiency for a given RoT constraint are enabled through such Node B controlled scheduling. Downlink signaling overhead is only required for a small number of scheduled UEs, rather than for all UEs in the case of a continuously updated TFCS. Furthermore, the scheduled mode can more precisely control how many UEs transmit data on their respective enhanced uplink channel in a given time interval. In the uplink of CDMA systems, simultaneous transmissions always interfere with each other and therefore, the scheduled mode can even ensure that always, for example, only one UE transmits data on its enhanced uplink channel at a time. Under certain conditions, this is likely to enhance throughput.

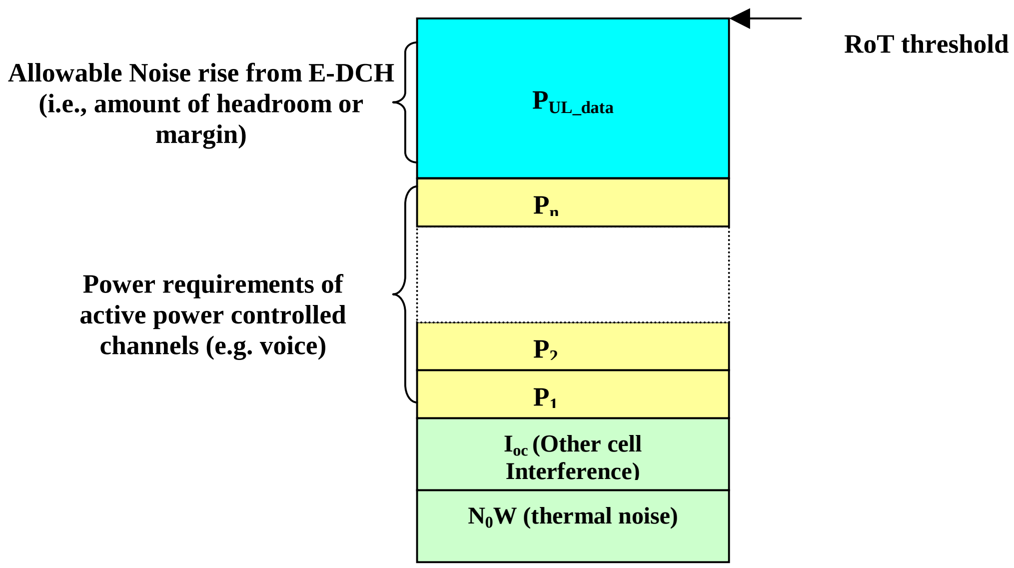

Figure 7.1.2: Noise Rise Bin for Node B controlled scheduling.

7.1.2.4 Issues Requiring Further Study

It is FFS how the method should work in soft handover. One problem is that scheduling UEs in soft handoff without any coordination between Node Bs in the active set could lead to RoT violations that significantly impact power controlled channels. However, one possibility is to simply send TFCS indicators that restrict UEs power level in soft handoff to control their interference impact on adjacent non-scheduling cells. The Node B would need to be made aware of a UEs soft handoff state in this case. Alternatively or additionally, TFC determination by the UE can include using soft handoff state information. Another limitation of scheduling a UE in soft handoff is that if the UE simply follows the scheduling command of either Node B, then the active set Node B(s) for the UE that do not schedule the user, may not attempt to decode its data. Therefore, the UE transmission will not derive any macro-diversity benefit. Yet another possiblility FFS is to use only TFCS control for UEs during soft handoff and allow autonomous transmissions. This alternative may avoid the complexity that could result in the operation of the Time and Rate scheduling in SHO. Finally, it is possible that each active set serving cell uses its knowledge of link imbalance (e.g. based on uplink

[1] Note that power status is also effectively updated at the serving Node B(s) by each uplink data transmission from the accompanying TFCI or TFRI information. It also may be advantageous to include buffer occupancy updates at the time of each uplink transmission in addition to periodic or triggered updates.

[2] Note that UE maximum power capability along with knowledge of the UE DPCCH power can be used for determining the TFCS indicator. Equivalently, Ec/Nt for the DPCCH measured at the Node B along with UE power margin to DPCCH power ratio can be used for determining the TFCS indicator.

DPCCH SNR consistently below the RNC defined outer loop power control threshold) to help limit scheduling activities for a given UE in soft handoff.

It is also FFS to minimize the number of scheduling information status update messages that are sent or alternatively how often scheduling information requests are made. Similarly, it needs to be determined whether UEs should autonomously report scheduling information (periodically and/or triggered on events) or whether they should only be requested by the Node B.

Finally, it is also for FFS on how to support both TFCS controlled autonomous transmissions and TFCS controlled and transmission time controlled scheduling for both the enhanced uplink DCH and along with the Rel’99/Rel’4/Rel’5 DCHs. The co-existence of the different modes may provide flexibility in serving the different traffic types. For example, traffic with small amount of data and/or higher priority such as TCP ACK may be sent using only a rate control mode with autonomous transmissions compared to using time and rate-control scheduling as the former would involve lower latency and lower signaling overhead. It also may be desirable to confine autonomous transmissions to specific time intervals different than when scheduled transmissions occur.

7.1.2.5 Signalling to Support Fast Node-B Time and Rate Control

Editor's Note: This chapter describes one possible solution for signalling to support the method.

7.1.2.5.1 L1 Signalling

Two new L1 messages are introduced in order to enable fast time and rate-control between the Node B and the UE.

- Scheduling Information Update (SI), sent in the uplink by the UE to the Node B. With the SI the UE can provide the Node B buffer occupancy and rate or power information so its scheduler(s) can maintain fairness and determine the UEs TFCS indicator and appropriate transmission time interval.
- Scheduling Assignment or Grant (SA), sent in the downlink by the Node B to the UE. With SA, the Node B can set the TFCS indicator and subsequent transmission start time(s) and time interval(s) to be used by the UE.

7.1.2.5.2 RRC Signalling (TBD)

7.1.2.5.3 Iub/Iur Signalling (TBD)

7.1.3 Scheduling in Soft Handover

When more than one Node B control the cells present in the UE active set, there are several alternatives as to the location of the scheduling entity which controls the UE. Possible solutions are:

- The Node B controlling the best downlink cell (as defined by RRC for DSCH/HS-DSCH operation) is identified as the sole scheduling entity.
- The Node B controlling the best uplink cell (the meaning of best uplink cell would have to be defined precisely) is identified as the sole scheduling entity for the UE.
- All Node Bs controlling one or more cells in the UE active set are identified as valid scheduling entities. This approach requires an additional decision procedure in the UE when the UE receives the scheduling assignments from multiple Node Bs.

If multiple Node Bs are identified as valid controlling entities, a UE in a SHO region may receive different scheduling assignments from multiple Node Bs and hence UE operation upon receiving the scheduling assignments should be defined. Possible UE operations are as follows:

- UE chooses the scheduling assignment from the ones indicated by the controlling Node Bs. For example, either the best scheduling assignment or the worst one can be chosen.
- UE combines the scheduling assignments from the controlling Node Bs based on a certain algorithm. For example, UE generates a single scheduling assignment by applying weighting factor (determined by the network) to each scheduling assignment.

Various options have to be considered in terms of system performance in particular in presence of link imbalance and in terms of overall system complexity. Reliability of downlink signalling in soft handover, e.g., the scheduling assignment(s) from the controlling Node B(s), should be taken into account in further evaluation.

If the Node B controlled scheduling in soft handover is not seen as feasible, then one possibility would be to turn off the Node B controlled E-DCH scheduling in soft handover.

7.2 Hybrid ARQ

7.2.1 General

Node B controlled hybrid ARQ allows for rapid retransmissions of erroneously received data units, thus reducing the number of RLC retransmissions and the associated delays. This can improve the quality of service experienced by the end user. As a Node B controlled retransmission is less costly from a delay perspective, the physical channel can be operated with somewhat higher error probability than in Rel 5, which may result in improved system capacity. The retransmission probability for the initial transmission is preferably in the order of 10-20% when evaluating hybrid ARQ as closed loop power control is used for the uplink, maintaining a given quality level. Significantly higher retransmission probabilities may lead to considerably reduced end user throughput, while at very small retransmission probabilities the Node B controlled hybrid ARQ will not provide any additional gains compared to R99/4/5. Soft combining can further improve the performance of a Node B controlled hybrid ARQ mechanism.

Not all services may allow for retransmissions, e.g., conversational services with strict delay requirements. Hybrid ARQ is thus mainly applicable to interactive and background services and, to some extent, to streaming services.

Thus, the major targets from a performance point of view with hybrid ARQ to consider in the evaluation of uplink hybrid ARQ are

- reduced delay
- increased user and system throughput

The design of an uplink hybrid ARQ scheme should take the following aspects into account:

- Memory requirements, both in the UE and the Node B. Rapid retransmissions reduce the amount of buffer memory required in the Node B for buffering of soft bits when a retransmission has been requested.
- Low overhead. The overhead in terms of power and number of bits required for the operation of the hybrid ARQ protocol should be low, both in uplink and downlink. Note that, unlike the HS-DSCH, the number of simultaneous users employing hybrid ARQ for transmitting data in the uplink may be significant, stressing the fact that the overhead for each user needs to be kept at a minimum.
- In-sequence delivery. The RLC requires in sequence delivery of MAC-d PDUs. Note that the in sequence delivery mechanism can be located either in the Node B or the RNC, depending on the scheme considered.
- Operation in soft handover. In soft handover, data is received by multiple Node Bs and alignment of a user's protocol state among different Node Bs needs to be considered. This problem is not present for the HS-DSCH, were reception occurs at a single node, the UE. Therefore, the feasibility of different modes of hybrid ARQ in conjunction with soft handover needs to be studied and, if found feasible, the cost of the required signaling investigated.
- Multiplexing of multiple transport channels. Hybrid ARQ cannot be used by all transport channels and multiplexing of transport channels using hybrid ARQ and those not using hybrid ARQ needs to be considered. In the downlink, there is a separate CCTrCh carrying the HS-DSCH, while the assumption of a separate CCTrCh is not necessarily true in the uplink scenario. In R99/4/5, only a single uplink CCTrCh is allowed.
- UE power limitations. The operation of the UE controlled TFC selection for R99/4/5 channels need to be taken into account in the design. In particular, UE power limitations in conjunction with activity on other transport channels with higher priority should be considered.
- Complexity. The hybrid ARQ schemes studied should minimize as much as possible the additional implementation complexity at all involved entities.

7.2.2 Transport Channel Processing

A protocol structure with multiple stop-and-wait hybrid ARQ processes can be used, similar to the scheme employed for the downlink HS-DSCH, but with appropriate modifications motivated by the differences between uplink and downlink. The use of hybrid ARQ affects multiple layers: the coding and soft combining/decoding is handled by the physical layer, while the retransmission protocol is handled by a new MAC entity located in the Node B and a corresponding entity located in the UE.

ACK/NAK signaling and retransmissions are done per uplink TTI basis. Whether multiple transport channels using hybrid ARQ are supported and whether there may be multiple transport blocks per TTI or not are to be studied further. The decision involves e.g. further discussion whether the current definition of handling logical channel priorities by the UE in the TFC selection algorithm remains as in R99/4/5 or if it is altered. It also involves a discussion on whether different priorities are allowed in the same TTI or not. The R99/4/5 specifications require a UE to maximize the transmission of highest priority logical channel in each TTI. If this rule is maintained, the delay for different logical channel priorities could be different, depending on whether the TFCS contains one or several transport channels.

Channel coding can be done in a similar way as in the R99/4/5 uplink. Transport blocks are coded and rate matching is used to match the number of coded bits to the number of channel bits. If multiple transport channels are multiplexed, rate matching will also be used to balance the quality requirements between the different transport channels. Note that multiplexing of several transport channels implies that the number of bits may vary between retransmissions depending on the activity, i.e., the retransmission may not necessarily consist of the same set of coded bits as the original transmission.

Unlike the downlink, the uplink is not code limited and initial transmissions typically use a lower code rate than is the case for HS-DSCH. Incremental redundancy with multiple redundancy versions is mainly beneficial at a relatively high initial code rate. Thus, the need for support of multiple redundancy versions may be smaller in the uplink than for the HS-DSCH. Explicit support for multiple redundancy versions, if desired, can be incorporated in the rate matching process as was done for HS-DSCH.

7.2.3 Associated Signaling

Associated control signaling required for the operation a particular scheme consists of downlink and uplink signaling. Different proposals may have different requirements on the necessary signaling. Furthermore, the signaling structure may depend on other uplink enhancements considered.

The overhead required should be kept small in order not to waste power and code resources in the downlink and not to create unnecessary interference in the uplink.

Downlink signaling consists of a single ACK/NAK per (uplink) TTI from the Node B. Similar to the HS-DSCH, a well-defined processing time from the reception of a transport block at the Node B to the transmission of the ACK/NAK in the downlink can be used in order to avoid explicit signaling of the hybrid ARQ process number along with the ACK/NAK. The details on how to transmit the ACK/NAK are to be studied further.

The necessary information needed by the Node B to operate the hybrid ARQ mechanism can be grouped into two different categories: information required prior to soft combining/decoding (outband signaling), and information required after successful decoding (inband signaling). Depending on the scheme considered, parts of the information might either be explicitly signaled or implicitly deduced, e.g., from CFN or SFN.

The information required prior to soft combining consists of:

- Hybrid ARQ process number.
- New data indicator. The new data indicator is used to control when the soft combining buffer should be cleared in the same way as for the HS-DSCH.
- Redundancy version. If multiple redundancy versions are supported, the redundancy version needs to be known to the Node B. The potential gains with explicit support of multiple redundancy versions should be carefully weighted against the increase in overhead due to the required signaling. Note that, unlike the HS-DSCH, the number of users simultaneously transmitting data in the uplink using hybrid ARQ may be significant.
- Rate matching parameters (number of physical channel bits, transport block size). This information is required for successful decoding. In R99/4/5, there is a one-to-one mapping between the number of physical channel

bits and the transport block size, given by the TFCI and attributes set by higher layer signaling. This assumption does not hold for hybrid ARQ schemes if the number of available channel bits varies between (re)transmissions, e.g., due to multiplexing with other transport channels. Hence, individual knowledge of these two quantities is required in the Node B.

The information required after successful decoding can be sent as a MAC header. The content is similar to the MAC-hs header, e.g., information for reordering, de-multiplexing of MAC-d PDUs, etc.

The information needed by UE necessary to operate the hybrid ARQ mechanism is either explicitly signaled by Node B, or decided by the UE itself, depending on the scheme. It is noted that whether the UE will decide the parameter values or the Node B will signal them, could affect the round trip time for HARQ retransmissions.

7.2.4 Operation in Soft Handover

The support of hybrid ARQ in different forms in soft handover requires careful consideration. In one possible scheme, all Node Bs serving the UE process the received data and transmit ACK or NAK to signal the result. If the UE does not receive an ACK from any of the involved Node Bs, it will schedule a retransmission. Otherwise, the transport block(s) will be considered as successfully transmitted and the UE will increment the new data indicator to signal to all involved Node Bs that the new data should not be soft combined with previous transmissions. To ensure that all involved Node Bs have the possibility to decode the transmission, regardless of the result from earlier transmissions, self-decodable transmissions are preferable.

A major problem with Node B controlled hybrid ARQ in soft handover is the link imbalance. Since the associated up- and downlink signaling does not benefit from the soft handover gain, it might be error-prone and/or require significant power offsets. Therefore, the feasibility of hybrid ARQ in soft handover situations should be investigated, taking the power required for control signaling into account. Protocol robustness in presence of signaling errors needs to be considered and additional protection of the control signaling may be required.

In the downlink direction, the UE may not be able to receive the ACK/NAK signals from all involved Node Bs. The consequences of downlink ACK/NAK errors are similar to the uplink ACK/NAK errors studied for HS-DSCH and it should be studied whether solutions similar to those used for HS-DSCH are applicable.

In the uplink direction, not all involved Node Bs may be able to receive the associated control signaling from the UE, which may lead to unsynchronised soft buffers between different Node Bs. This could result in erroneous combining of new packets with previously stored packets that have not been flushed. One possibility to reduce the occurrence of erroneous combining could be to increase the reliability of the uplink HARQ control signaling. This could be for example done by power offsets or by increasing the number of bits for the New Data Indicator thus making a wrap around of the NDI less likely. An alternative could be to operate without soft combining in soft handover situations, removing the need for reliable outband signaling of the new data indicator and the hybrid ARQ process number. More robust inband signaling can be used for these quantities instead. Node B controlled ARQ without soft combining could be considered in non-soft-handover as well, if clear gains are seen only from the ARQ mechanism and not from the soft combining itself. Another possibility, preserving support for hybrid ARQ with soft combining in soft handover, could be to synchronize the NodeB's soft buffer content via additional network signalling or to locate the soft buffer in the Node B and the final ACK/NAK decision in the RNC. This technique allows the RNC to align the soft buffer status in each Node B and may benefit from the soft handover gain for the related hybrid ARQ control signaling, but the delays will be larger than for a pure Node B controlled scheme.

7.3 Fast DCH Setup Mechanisms

7.3.1 Background

Possible enhancements include, but are not limited to, the physical layer random access procedures, NBAP/RRC signaling, and uplink/downlink synchronization procedures. Any enhancement, or combination of enhancements, to the procedures for fast DCH establishments should fulfill the following requirements:

- Allow for significant reduction in switching delays.
- Fit into the connection state model and, to the extent possible, reuse existing procedures and techniques.
- Allow for unaffected operation of existing UEs and Node Bs

7.3.2 Reducing Uplink/Downlink Synchronization Time

Establishing a DCH requires the UE and Node B to synchronize the physical up- and downlink channels as briefly described in Section 6.1.1. Techniques to reduce the downlink and/or uplink synchronization time should be studied as a part of the overall goal of reducing the delays associated with DCH establishment.

The overall delay from t_1 to t_7 in Figure 6.1.2 depends both on the implementation, the performance requirements on the UE, and the procedures in the 3GPP specifications. T_1 and T_2 mainly depend on network implementation. T_3 depends on the TTI used for FACH, which could be shortened at the cost of a reduced interleaving gain, and the UE processing delays. In this section, a technique for reducing T_4, accounting for 40-160 ms delay, and T_5, accounting for 10-70 ms delay, by using an improved synchronization scheme is proposed.

The proposed enhancement is illustrated in Figure 7.3.1. The basic idea is to replace the presently defined DPCCH uplink and downlink synchronization scheme requiring a time interval T_4+T_5 (specified in [14]) with an enhanced scheme reducing this time to 10 ms. A power ramping procedure is used, where the power of the uplink DPCCH is ramped up from a calculated initial power level by sending power up commands from the Node B until the Node B has obtained synchronization to the uplink signal. Acquisition of the uplink signal is indicated to the UE on the downlink DPCCH simply by sending power down commands. In the radio frame following the power control preamble, data transmission on both uplink and downlink DPDCH can start.

InFigure 7.3.2, the power ramping phase is illustrated in more detail. Downlink and uplink DPCH transmission shall start at the same frame number, which shall be indicated in the switching message to the UE. Note that the UE already has received data on the S-CCPCH and thus is synchronized to the network, and the relative timing between downlink DPCH and S-CCPCH is known from L3 signaling, In Figure 7.3.2, downlink transmission starts at time instant t_1 (which corresponds to $t_4 = t_5$ in Figure 7.3.1), with some offset relative to the frame timing of the CPICH. The offset is indicated to the UE in the switching command. Uplink transmission shall start with a timing offset relative to the downlink DPCH, i.e., at $t_1+T_0+\tau$, where τ is the delay of the first detected path measured on CPICH and T_0 = 1024 chip intervals, as specified in [14]

For uplink ramping, a predefined setting of all DPCCH bits is preferably used to make it possible to collect all transmitted energy for initial synchronization in the Node B receiver without caring on modulation. Uplink DPCCH power is ramped up with one step per slot. In the ramping phase, downlink TPC bits from the Node B should be set to "up". As soon as the Node B receiver has been reliably synchronized to the uplink, the Node B shall enter power control operation, i.e., transmit up/down power control commands and evaluate the TPC information received on the uplink DPCCH (time instant t_2 in Figure 7.3.2). In-sync detection is tested in Node B similarly as for PRACH preambles based on thresholds. The UE is informed when Node B obtains in-sync through the TPC pattern received on the downlink.

Note that the Node B uplink receiver can collect the energy for the entire ramping phase, not only the energy of the last slot. Furthermore, as there is no modulation present on the DPCCH, it is possible to achieve a very large processing gain at the receiver, equal to all 2560 chips (34 dB). This allows for very power efficient, highly secure detection of the DPCCH transmission in the Node B. One possibility is to use peak detection in long-term delay power spectrum estimations, which for instance can be calculated with a matched filter.

The initial downlink DPCCH power level is determined in the same fashion as in the present procedure, i.e., by using the initial downlink DPCH power level IE present in the "Radio Link Setup/Addition Request" messages. Setting of the initial power is implementation dependent. If prior information on the distance between UE and Node B or a path loss measurement is available in the RNC, this can be used for more tight setting of the initial downlink DPCCH power level. If no distance or path loss information is available, a "broadcast power level" needs to be employed. To secure reception of the downlink DPCCH, its initial power should in any case be chosen somewhat higher than needed according to pre-calculations. This means that as soon as the inner power control loop starts operation (time instant t_2 in Figure 7.3.2), it is very likely that downlink power is ramped down first. In the proposed fast synchronization scheme, setting of initial downlink power is much less critical than in the Rel99/4/5 scheme as a somewhat too high power would be employed only for a very short time interval.

DPCH setup failure in the Node B is identified when no uplink synchronization is obtained within the preamble period. In the case, the downlink DPCCH transmission should be stopped at the end of the preamble interval. Stop of downlink transmissions shall be identified in the UE by means of a fast DL DPCCH synchronization status detection scheme and stop further uplink transmissions. Further handling of DPCH setup failure could be done in several ways. For instance, a new attempt could be made a predefined time after the first try. Alternatively, the physical channel reconfiguration failure procedure as defined in [15]. could apply also for this new scheme.

Introducing enhancements such as those described above can be done by defining "Synchronization Procedure C" in addition to procedures A and B already specified in [14]. The impact on higher layers, the interaction with power control, and in which scenarios a new synchronization procedure may be applied are for further study.

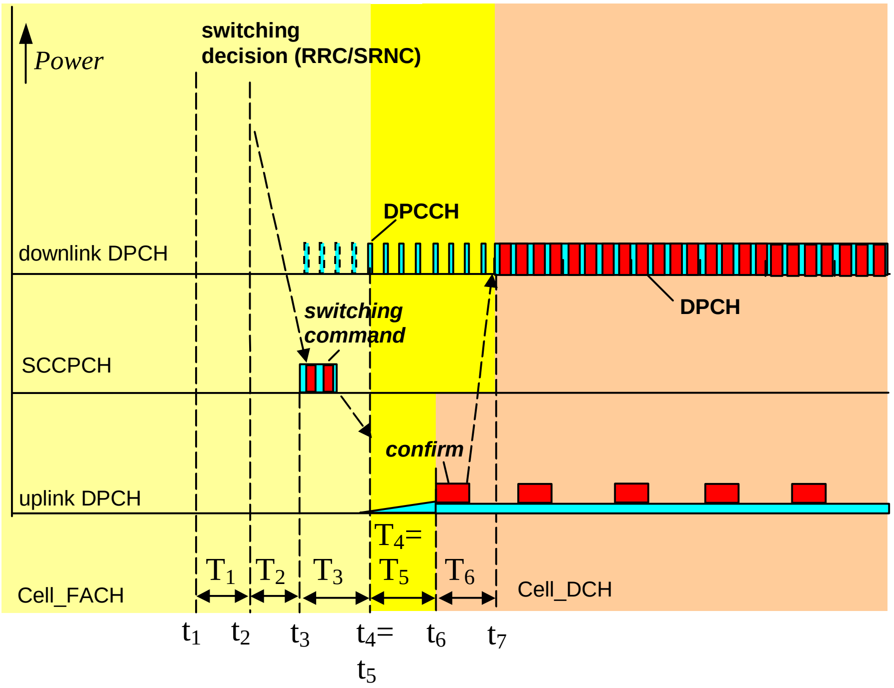

Figure 7.3.1: Enhanced procedure for DCH establishment.

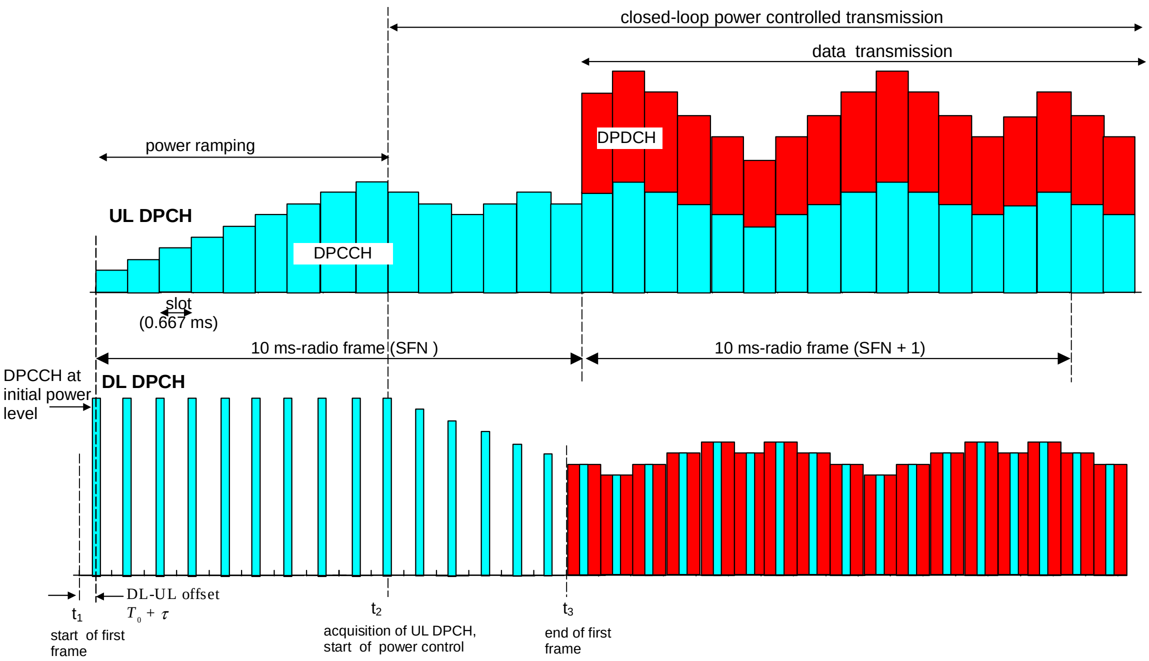

Figure 7.3.2: Illustration of the enhanced uplink/downlink synchronization scheme.

7.4 Shorter Frame Size for Improved QoS

Reducing the minimum TTI supported from the 10 ms in Rel5 to a lower value may reduce the transfer delay through a reduced Uu transfer delay and reduced delays due to TTI alignment (incoming data to be transmitted has to wait until the start of the next TTI). A reduced TTI may also allow for reduced processing time as the payload sizes are reduced compared to a larger TTI, a shortened roundtrip time in Node B controlled hybrid ARQ protocols and reduced latencies in some scheduling schemes. Reduced delays may also result in a higher system throughput and better resource utilization.

Thus, the major targets from a performance point of view with a reduced uplink TTI are:

- improved end-user quality
- increased user and system throughput
- significant delay reduction

The introduction of a reduced TTI should take the following aspects into account:

- End-user delay. Any reduced TTI considered should result in the possibility for a significant reduction in uplink delay while still support reasonable payloads.
- Choice of shorter TTI. It is preferable if the Rel5 minimum TTI of 10 ms is a multiple of the reduced TTI considered. The obvious choice is a 2 ms TTI, which also is an alignment to the short TTI adopted for HS-DSCH.
- Link performance. The influence of a short TTI on link performance need to be considered.
- Channel structure. Support of services and applications using Rel5 channels should be considered. The operation of UE controlled TFC selection need to be taken into account. Any increase in UE peak-to-average ratio should be analyzed and kept low.
- Complexity. Any complexity increase due to a reduced TTI should be clearly motivated by a corresponding performance gain.

7.5 Signalling to support the enhancements

Editor's Note: This section shall describe, what kind of new signalling the evaluated enhancement techniques require.

8 Physical Layer Structure Alternatives for Enhanced Uplink DCH

Editor's Note: This section is expected to contain a more detailed description of proposal physical layer structure(s) in time and code domain. This section will be used as a basis for defining the simulation assumptions in the annex. This chapter shall also describe the timing relationship of the new physical layer channels with respect to the Rel'5 physical layer channels.

8.1 Relationship to existing transport channels

It remains to be determined whether there will be a new transport channel added to RAN specification. Uplink enhancements may

- consist of methods limited on improving the utilization of existing transport channels or
- introduce methods that require new transport and physical channels

In order to encompass both possibilities, the transport channel is referred here as E-DCH.

8.2. TTI length vs. HARQ physical channel structure

Two different TTIs have been mentioned in conjunction with uplink enhancements: either reusing the existing R99 10 ms TTI or introducing a shorter (e.g., 2 ms) TTI:

- Using a 10 ms TTI allows for reusing the R99 DPDCH structure, including baseband processing and TFCI signaling. The drawback is the, compared to a shorter TTI, larger delays. Using QPSK in the uplink can lead to an increase in PAR, although the value of the PAR increase remains to be investigated.
- Using a 2 ms TTI allows for reduced delays. The drawback is the need for a new physical layer frame structure and TFCI-like signaling. The most straightforward way of supporting a short (2 ms) TTI seems to be the introduction of a new code multiplexed physical channel in the uplink. Using additional codes in the uplink can lead to an increase in PAR, although the value of the PAR increase remains to be investigated.

These TTI lengths of 10 ms and 2 ms are considered here as examples.

If E-DCH utilizes physical layer HARQ, there is a need to transmit ACK/NACK signaling in a downlink physical channel. N defines the minimum number of HARQ processes required to provide continuous transmission. However, increasing the number of HARQ channels also adds to round trip time and thus N cannot be arbitrarily large. A compromise between round trip time and processing time is of main importance when considering the selection of N.

If the available time for downlink signaling and UE/Node B processing is made long enough through suitable selection of N, ACK/NACK could be embedded in existing Rel'99 downlink channel structure, i.e. within a 10 ms TTI. Another option is to reserve a specific field shorter than 10 ms time period, in a downlink physical channel for ACK/NACK as is done in HS-DPCCH for uplink in Rel'5 HSDPA. The downlink ACK/NACK field length is naturally independent of TTI length in uplink.

Figure 8.2.1 depicts the general concept of timing for E-DCH HARQ process. After having received transport block(s) on E-DCH the Node B has T_{NBP} for processing and sending acknowledgement to the UE. In here no assumption is made on which downlink physical channel the ACK/NACK in DL would be sent. Based on the acknowledgement and possible other information provided by the UTRAN, the UE decides whether it resends the transport block(s) or transmits new transport block(s). The processing time available for the UE between receiving the acknowledgement and transmitting the next TTI in the same HARQ process is T_{UEP}.

The length of the acknowledgement field in DL directly affects the available processing time in Node B and UE. The length of the acknowledgement field might also affect the required power offset for transmitting it, relative to DL DPCCH, depending on the scheme. With 10 ms TTI and high enough N, acknowledgement could e,g, be embedded in existing multiplexing structure within a 10 ms TTI. This might allow more space for coding and smaller power offset for transmitting ACK/NACK than in the case where ACK/NACK is inserted into downlink physical channel within a shorter time period than 10ms.

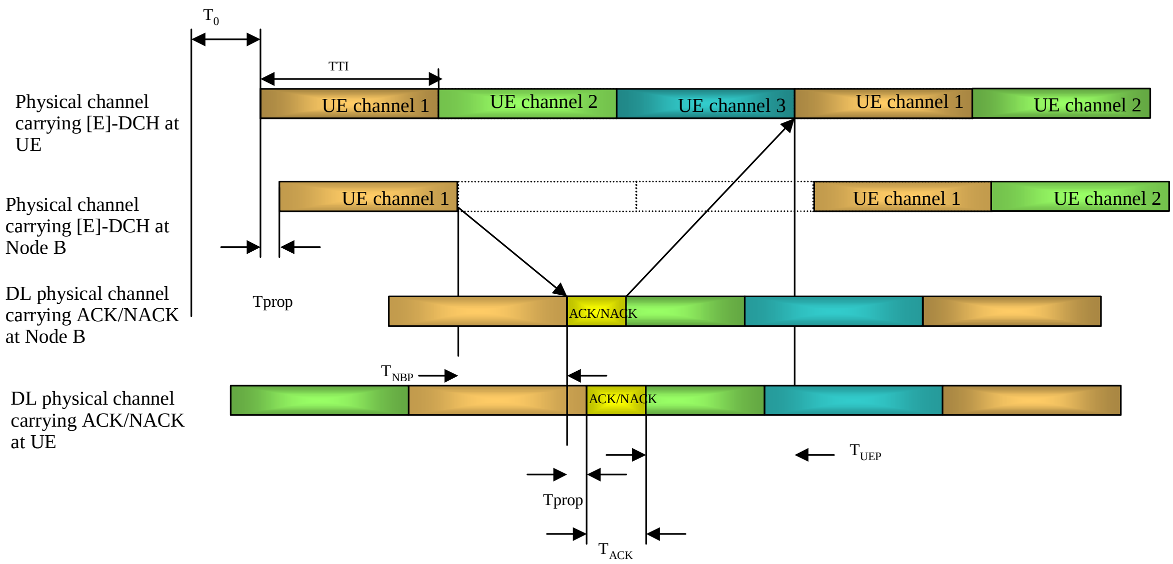

Figure 8.2.1. HARQ timing schematic for N=3, TTI=10 ms, as an example.

Table 8.2.1 presents some estimations for available processing time TTI lengths 10 ms and 2 ms, with N=2,3,4,5. The timing calculations assume a roundtrip delay of 0.1 ms. The acknowledgement signal from the Node B may be spread over one of more slots. However, the longer T_{ACK} becomes, the less processing time there is available for UE and RNS. For TTI=10 ms case, a T_{ACK} = 10 ms is possible if N=3 or larger. With TTI=2 ms, T_{ACK} necessarily has to be shorter.

Table 8.2.1. Examples of UE and Node B processing times with E-DCH

TTI length (ms)	N	T_{ack} (ms)	$T_{NBP}+T_{UEP}$ (ms)
10	2	2	7.9 (0.8xTTI)
10	3	2	17.9 (1.8xTTI)
10	3	10	9.9 (1.0xTTI)
10	4	2	27.9 (2.8xTTI)
10	4	10	19.9 (2.0xTTI)
2	4	2 (3 slots)	3.9 (1.95 x TTI)
2	5	2/3 (1 slot)	7.23 (3.6xTTI)
2	5	4/3 (2 slots)	6.56 (3.3xTTI)
2	6	4/3 (2 slots)	8.56 (4.3xTTI)
2	6	2 (3 slots)	7.90 (4.0xTTI)
2	7	2 (3 slots)	9.90 (5.0xTTI)

The table shows examples of the total time available for UE and Node B processing in the case of implicit scheduling. Thus, the figures in Table 8.2.1 represent minimum round trip time. Other methods with e.g. additional control channels would increase the round trip time or reduce available processing time. These methods are investigated separately. Note that the length of the E-DCH TTI also has an impact on the processing time needed. Since a shorter TTI contains fewer bits than a longer one, the processing load for baseband processing such as interleaving and turbo decoding is smaller and less time is consumed. On the other hand, interleaving gain is impacted when short TTI length is employed.

The choice of TTI and N should be done in conjunction with selecting the structure of the downlink ACK/NAK transmission. Furthermore, the maximum data rate supported will affect the required processing times. Herein, the assumption that maximum data rate would be around 1-2Mbit/s was used.

More detailed analysis of the required processing times are needed in the future, but this gives some rough estimate how the TTI length affects the HARQ physical layer structure. In addition to processing times, important issues to consider are the physical layer structure for sending the L1 signaling in uplink and downlink, and the performance and complexity related to that.

8.3 Multiplexing alternatives

This chapter is describing the different alternatives of how E-DCH can be multiplexed with the existing Rel'99 channel structures. (E-DCH is used as a general term referring to both a possible new type of transport channel and to possible enhancements to an existing transport channel)

There are basically two different alternatives to introduce the E-DCH: it can either be time multiplexed with other DCHs in the same way as different DCHs are multiplexed in Rel'99 or it can be code multiplexed, i.e., sent using a dedicated code channel. These alternatives are described and discussed in the following subsections.

Issues that need to be studied when considering each multiplexing alternative are:

- Possible introduction of TTI lengths shorter than 10ms
- Possible Slot or frame synchronism for E-DCH users
- Flexibility of H-ARQ operation for both soft-handoff and non soft-handoff case.
- Variable gain factors and modulation for E-DCH
- Peak to Average Power Ratio (PAR)
- Interoperability with Rel’99/Rel’4/Rel’5 base stations and support of existing R99/4/5 channels

- Impact of possible introduction of shorter TTI lengths to TFC selection algorithm
- Impact of possible introduction of multiple uplink CCTrCHs to higher layers

8.3.1 Reuse of current physical layer structure

In this alternative the E-DCH is time multiplexed into the same coded composite transport channel (CCTrCH) as the other DCHs if present. The TFCI indirectly informs where and how many bits of each DCH within the CCTrCH are, regardless of the DCH being a Rel99 DCH or an E-DCH.

Time multiplexing is easiest to implement if the TTI length is 10/20/40/80 ms, since then the Rel'99 transport channel multiplexing chain can be used. There may naturally be some enhancements, e.g., to rate matching, to support the potential new enhanced uplink features, e.g., hybrid ARQ.

The advantage of time multiplexing of the E-DCH with Rel99 DCHs is that no new code channels are unnecessarily introduced. The multicode transmission would only be used for high data rates in a similar way as specified in Rel99. This approach minimises the required peak to average power ratio (PAR) in the UE transmitter provided only one DPDCH is used. The code channel structure of this alternative is the same than is already used in Rel'99.

It may be difficult to use higher order modulation and variable gain factors with this approach. Further, the number of available channel bits on a DPDCH for E-DCH depends on the presence of higher priority DCH's (e.g voice) and may impact the flexibility of HARQ operation.

8.3.2 Allocating a separate code channel for Enhanced uplink DCH

In this alternative the E-DCH is code multiplexed with other DCHs, i.e., sent using a dedicated code channel, thus introducing a new CCTrCH in the uplink. (Note, that Rel'99 only allows one CCTrCH in the uplink per UE.)

The advantages of code-multiplexing include, among others, simpler introduction of new/shorter TTI lengths, increased flexibility of HARQ operation , and support of adaptive modulation.

Introducing a new code channel may increase PAR in some cases which should be studied. Further, the available resources, such as power, for the code channel carrying E-DCH depends on the presence of higher priority DCHs being carried on the other code channels.

9 Evaluation of Techniques for Enhanced Uplink

Editor's Note: In this chapter, the techniques that are expected to provide potential gain are evaluated in more detail, both from performance and complexity point of view. Also the backwards compatibility with the features introduced in the previous versions of the 3GPP specifications are to be considered keeping in mind the gain versus complexity issue. E.g. chapters should clarify, should the new feature function in soft handover, with the relating complexity aspects, should possible new downlink channels support all transmit diversity modes, etc.

9.1 Scheduling <NodeB controlled scheduling, AMC>

9.1.1 Performance Evaluation

9.1.2 Complexity Evaluation <UE and RNS impacts>

9.1.3 Downlink Signalling

9.1.4 Uplink Signalling

9.2 Hybrid ARQ

9.2.1 Performance Evaluation

9.2.1.1 Hybrid ARQ performance with and without soft combining

In this section, link level performance results of the hybrid ARQ with and without chase combining are presented for 144 kbps and 480 kbps with the Rel-99 turbo code of 1/3 coding rate and the Rel-99 rate matching. The results are provided on ITU Pedestrian A channel at 3kmph and 30kmph.

Simulation assumptions are listed in the Table 9.2.1 below.

Table 9.2.1. Simulation assumptions

Chip Rate	3.840 Mcps
Carrier Frequency	2 GHz
Propagation Channel	Pedestrian A 3km/h, 30km/h
Channel Estimation (CE)	real CE (DPCCH 6 pilot bits)
Inner-loop transmit power control (TPC)	On
Outer-loop power control	Off
TPC step size	1dB
TPC delay and error rate	1 slot, 4%
Receiver	Rake
Antenna configuration	2 antenna space diversity
Channel oversampling	1 sample/chip
Turbo code information	R=1/3, K=4, 8 iteration, Decoder : Max Log MAP
Information bit rate	144kbps / 480kbps
SF	8 (144kbps) / 4 (480kbps)
Modulation	Dual BPSK
E-DCH TTI	10ms
Hybrid ARQ	Chase Combining(CC) / No Combining(NC)
Maximum number of transmission	3
ACK/NACK signaling error	No error
Rate matching	Rel'99 Rate matching
Gain factor	β_c =5, $\beta_{E-DPDCH}$ =15
Cell configuration	Single omni-cell and single user

The throughput is calculated as

$$Throughput = \frac{R_{\inf}}{N_{av}}$$

where $R_{\inf}$ is the information bit rate and N_{av} is average number of transmissions.

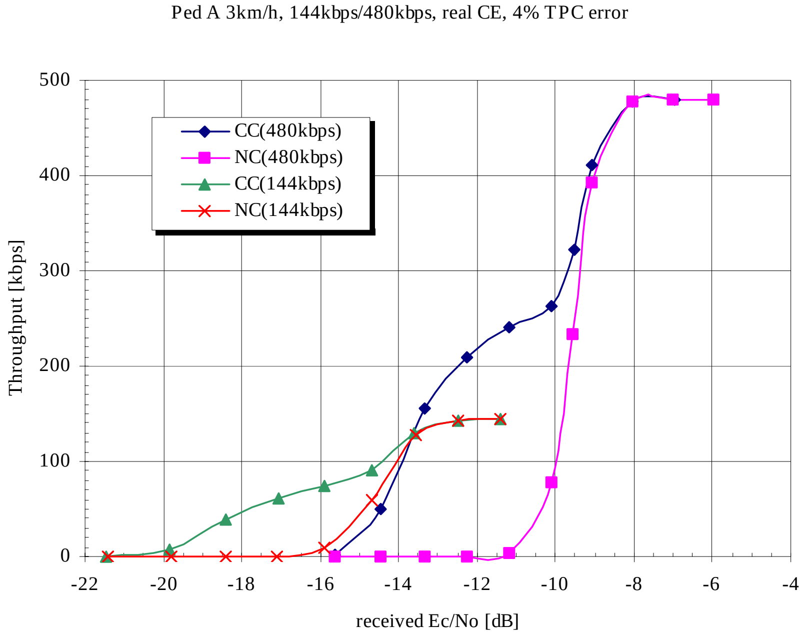

Figure 9.2.1. Throughput in Pedestrian A 3 km/h with power control

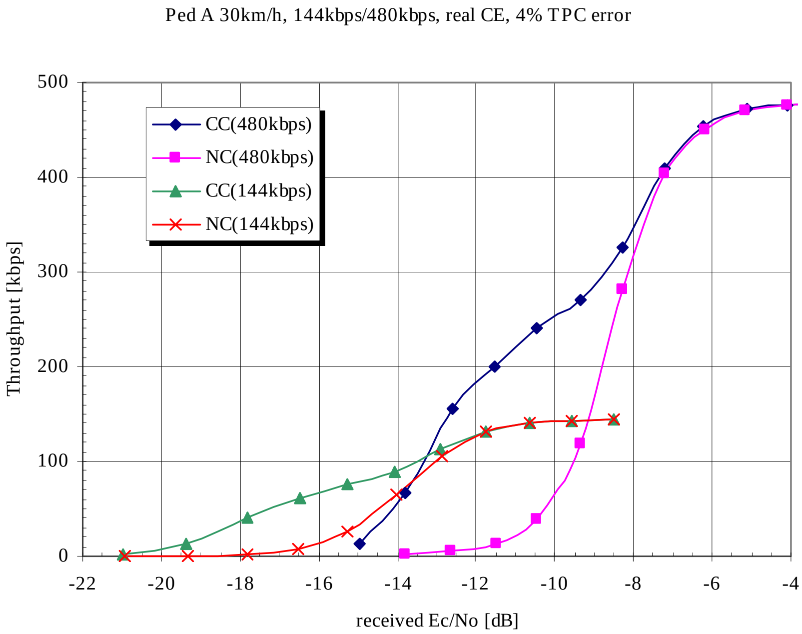

Figure 9.2.2. Throughput in Pedestrian A 30 km/h with power control

Figure 9.2.1 and Figure 9.2.2 show the throughput performance in Pedestrian A with 3 km/h and 30km/h, respectively. It can be seen that the chase combining provides throughput gain when the UE available power is limited so that the hybrid ARQ without chase combining suffers from throughput loss. It is noted that the gain from the soft combining in a realistic scenario should be studied further, since more than two data rates could be typically available to choose depending on, e.g., the UE available power and the scheduling command received from the scheduling Node B(s).

Figure 9.2.3 and Figure 9.2.4 show the average number of transmissions in Pedestrian A 3 km/h and 30km/h, respectively. It can be seen that the chase combining can reduce the number of transmissions significantly.

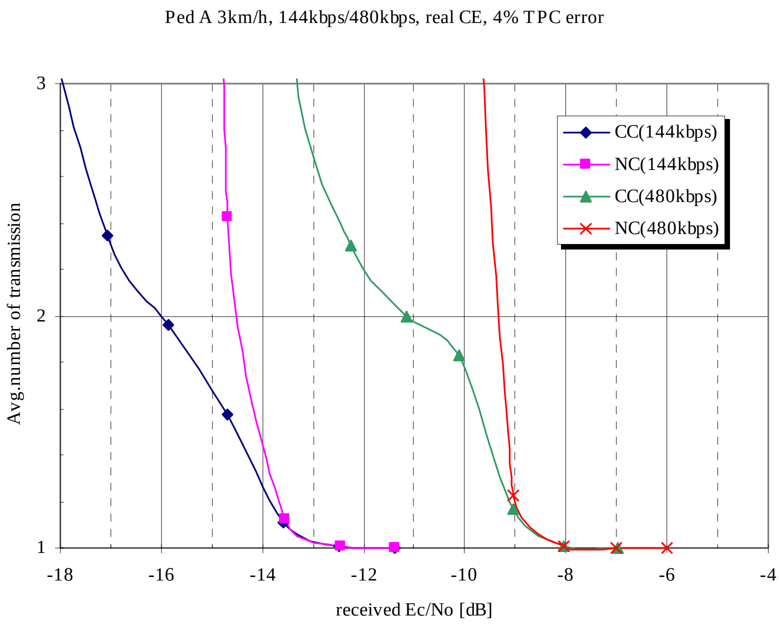

Figure 9.2.3. Average number of transmissions in Pedestrian A 3km/h with power control

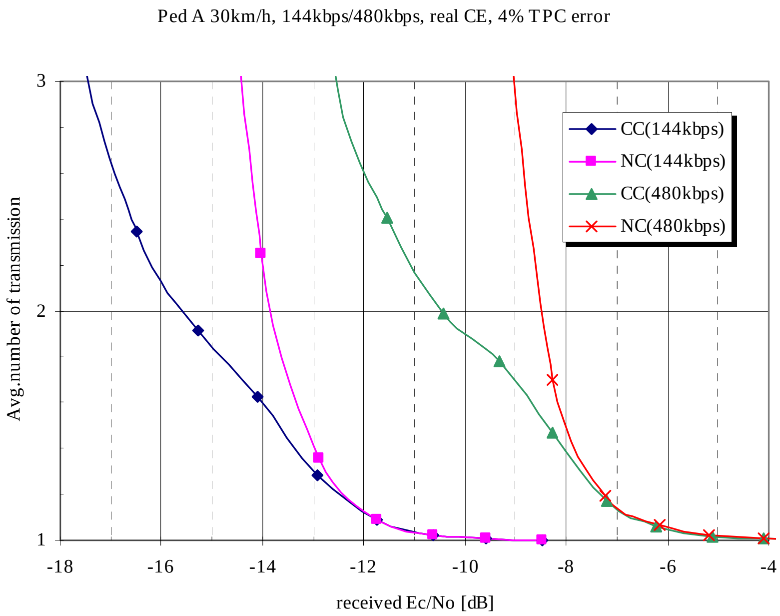

Figure 9.2.4. Average number of transmissions in Pedestrian A 30km/h with power control

Figure 9.2.5 shows the BLER curve of 480 kbps in Pedestrian A 3km/h for each transmission with the chase combining.

Figure 9.2.6 and Figure 9.2.7 show the delay distributions with the first transmission BLER = 17% and 49%, respectively. It can be seen that the chase combining can cut down the number of transmissions to two transmissions even with the first transmission BLER = 49%. The gain of the chase combining in delay distribution is more emphasized with the higher first transmission BLER. This could be beneficial especially for delay sensitive applications.

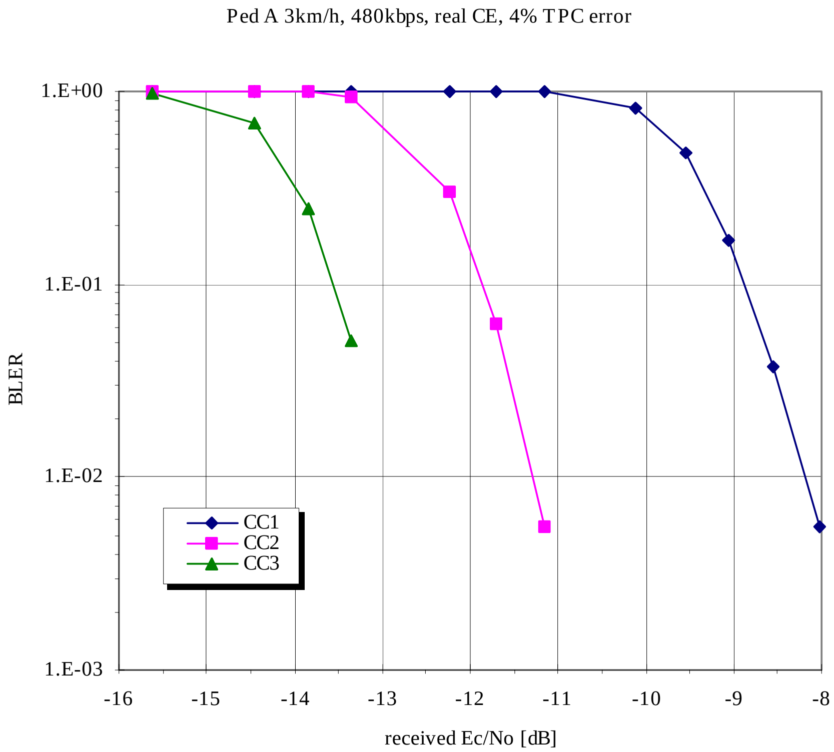

Figure 9.2.5. BLER for 480 kbps in Pedestrian A 3km/h

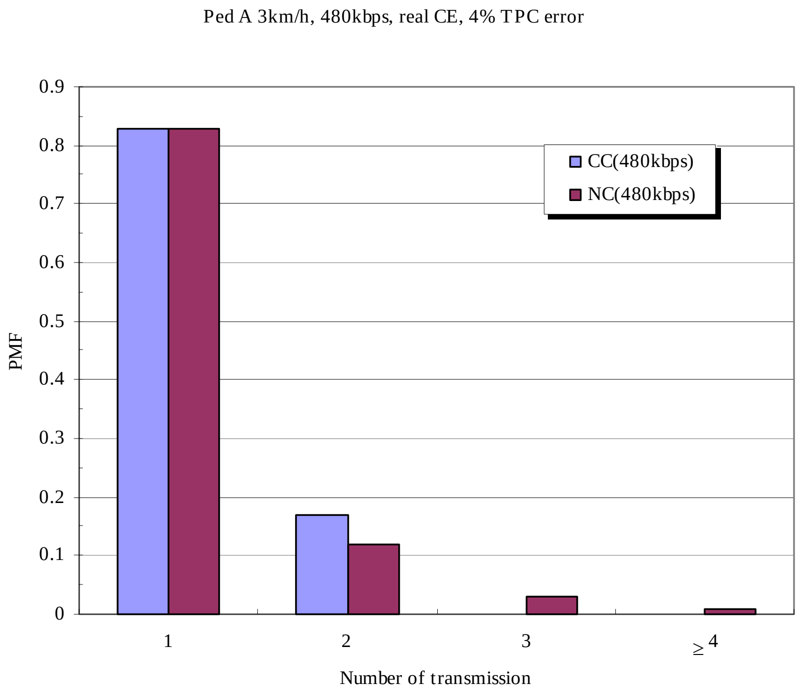

Figure 9.2.6. Delay distribution with the first transmission BLER = 17% for 480 kbps in Pedestrian A 3km/h

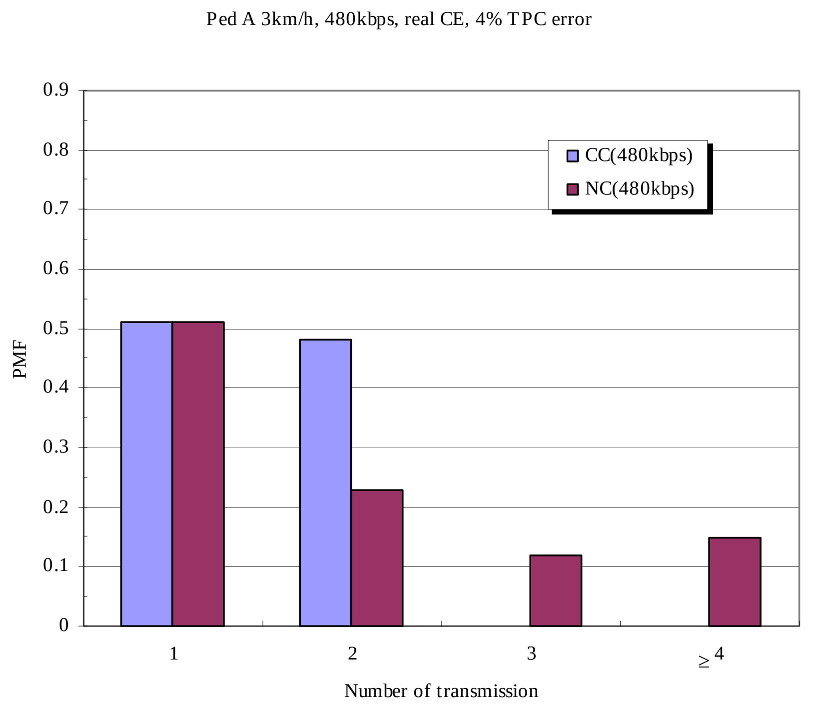

Figure 9.2.7. Delay distribution with the first transmission BLER = 49% for 480 kbps in Pedestrian A 3km/h

9.2.1.2 HARQ Efficiency

In this section, the benefits of link level retransmissions, and issues of HARQ efficiency and the maximum number of retransmissions needed to support on E-DPDCH are addressed. Here, E-DPDCH denotes the physical set of channelization codes used to carry E-DCH content.

A reference MCS for a sample 2ms TTI is shown in Table 9.2.1.2.1.

Index	Transport Block Size	Mod	Code Rate	Rate (kbps)		
				1 Tx	2 Tx	4 Tx
4	1280	QPSK	0.333	640	320	160
7	2048	QPSK	0.533	1024	512	256
9	2560	QPSK	0.333	1280	640	320
15	4096	QPSK	0.533	2048	1024	512
19	5120	QPSK	0.444	2560	1280	640
31	8192	QPSK	0.711	4096	2048	1024

Table 9.2.1.2.1 Reference MCS – 2ms TTI

From Table 9.2.1.2.1, the same target rate can be achieved using different transport formats and number of transmissions. The performance of E-DPDCH is now evaluated with 1 or 2 or 4 target transmissions for the *same* target data rate, as shown in Table 9.2.1.2.2.

Target Data Rate (kbps)	**MCS**		
	1 Tx	**2 Tx**	**4 Tx**
640	4	9	19
1024	7	15	31

Table 9.2.1.2.2 Simulation Set – 2ms TTI

The simulation assumptions and results are shown in Annex A.2.1.1.

For a 2ms sample TTI and associated link level performance shown in Figures A.2.1.1.1 to A.2.1.1.4 and Tables A.2.1.1.1 and A.2.1.1.2, it is seen that:

1. For the same *target* data rate, as the *target* number of transmissions increases, the link efficiency improves.
   a. The efficiency improvement reduces as the base number of transmissions increases.
   b. The link efficiency gain from 1 to 2 transmissions is more than the gain from 2 to 4 transmissions.
2. The optimal DPCCH SNR typically decreases and E-DPDCH/DPCCH power ratio increases as the number of transmissions increases.
3. For the same *target* data rate, as the maximum number of transmissions increases, the E-DPDCH can be terminated relatively earlier ➔ effective data rate is higher.
   a. For a target *maximum* of 2 transmissions, the average number of *required* transmissions is 1.7 ➔ early termination factor is 1.7/2 = 0.85
   b. For a target *maximum* of 4 transmissions, the average number of required transmissions is 3.0 ➔ early termination factor is 3.0/4 = 0.75
4. For the same *effective* data rate, as the maximum number of transmissions increases, the link efficiency increases.
5. The first transmission BLER can be very high for the most efficient link operation
   a. This does not necessarily maximize throughput.
6. For the same number of target transmissions, throughput can be maximized or delay can be reduced, at the cost of link efficiency.
7. For the same effective data rate and same base TTI, as the maximum number of transmissions increases, the average delay increases.

9.2.2 Complexity Evaluation <UE and RNS impacts>

9.2.3 Downlink Signalling

9.2.4 Uplink Signalling

9.3 Fast DCH Setup Mechanisms

9.3.1 Performance Evaluation

9.3.2 Complexity Evaluation <UE and RNS impacts>

9.3.3 Downlink Signalling

9.3.4 Uplink Signalling

9.4 Shorter Frame Size for Improved QoS

9.4.1 Performance Evaluation

9.4.2 Complexity Evaluation <UE and RNS impacts>

9.4.3 Downlink Signalling

9.4.4 Uplink Signalling

10 Impacts to the Radio Network Protocol Architecture

Editor's Note: Input from RAN2 is expected for this chapter

11 Impacts to L2/L3 Protocols

Editor's Note: Input from RAN2 and RAN3 is expected for this chapter

12 Conclusions and Recommendations

Annex A: Simulation Assumptions and Results

A.1 Link Simulation Assumptions

A.1.1 Interface between link level and system level

The performance characteristics of individual links used in system simulation are generated a priori from link level simulations. Due to weak uplink pilots, and the resulting poor channel estimates, the link performance predicted by methods that do not account for imperfect channel estimates can yield incorrect results. So, it is very important to account for the effect of channel estimation errors on link performance. Suggested techniques for predicting link error performance in the presence of channel estimation errors are discussed further in the Annex E.

In general, there are two cases of interest:

1. *Comparison of techniques/proposals without H-ARQ*: If the effect of channel estimation errors on link performance is modeled in generating simulations results for a comparison of techniques without H-ARQ, the link error prediction method used should be stated in the simulation assumptions. Otherwise, justification should be provided as to why the comparison is valid. (See Annex E.)

2. *All other cases, i.e., comparison of techniques, one or all of which include H-ARQ combining*: In all these cases, the effect of channel estimation errors on link performance must be accounted for in generating simulation results for the comparison. (See Annex E)

The following table should be included along with the simulation assumptions accompanying all results:

Are any of the techniques being simulated involve H-ARQ combining?	**Is the effect of channel estimation errors on link performance accounted for?**	**Comments**
Yes/No	Yes/No	If the effect of channel estimation errors on link performance is modeled, then state the method. Otherwise, justify why the comparison is valid.

A.1.2 Link level parameters

Table A - 1 below shows the general link level parameters, to be used both in the reference case, and in the new schemes proposed for Enhanced Uplink DCH. Table A - 2 shows the link level parameters to be used in the reference case.

Table A - 1 - General link level parameters

Parameter	Explanation/Assumption	Comments
Channel coder	Turbo 1/3	
Number of iterations for turbo decoder	8	
Turbo decoder	Max Log MAP	
Channel models/ UE speed for channel model	Pedestrian B / 3 km/h, Vehicular A / 30 km/h Pedestrian A / 3 km/h Optional Vehicular A / 120kph	One channel model per simulation

Table A - 2 - Link level parameters for the Rel99/Rel4/Rel5 reference case

Parameter	Explanation/Assumption	Comments
CL power control	ON	
CL power control error rate	4%	
TTI	10 ms	
User data rates in TFCS	8, 16, 32, 64, 128, 256, 384 kbit/s	These data rates are included in the TFC selection modelling in the system level.

A.1.3 Channel models

ITU channel models [2] are used in the link level and system level simulations. Multipath intensity profiles are given below.

The multipath intensity profile of the Pedestrian-A channel is defined as follows:

Relative Delay (ns)	0	110	190	410
Relative Power (dB)	0.0	-9.7	-19.2	-22.8

Table A - 3 - ITU Pedestrian-A channel model.

The multipath intensity profile of the Pedestrian-B channel is defined as follows:

Relative Delay (ns)	0	200	800	1200	2300	3700
Relative Power (dB)	0.0	-0.9	-4.9	-8.0	-7.8	-23.9

Table A - 4 – ITU Pedestrian-B channel model

The multipath intensity profile of the Vehicular-A channel is defined as follows:

Relative Delay (ns)	0	310	710	1090	1730	2510

Relative Power (dB)	0.0	-1.0	-9.0	-10.0	-15.0	-20.0

Table A - 5 – ITU Vehicular-A channel model

The delay intensity profiles are computed from the ITU channel multipath intensity profiles given in the Tables above for a set of transmit and receive filters. The delay intensity profile for 5MHz WCDMA transmit and receive filters (raised cosine with beta=0.22) for a chip rate of 3.84Mcps are given in Table A - 6 The Fractional Recovered Power (FRP) is given in Table A - 6 for each recovered ray. Fraction of un-Recovered Power (FURP) shall contribute to the interference of the finger demodulator outputs as an independent fader.

Table A - 6 - FRP and Delay profile for each ITU channel model for 5MHz bandwidth and 3.84Mcps.

Multi-path	FRP for each ray (dB)					Delay for each ray (Tc)				
Model	1	2	3	4	5	1	2	3	4	5
Pedestrian A	-0.22					0.125				
Pedestrian B	-3.39	-8.63	-8.45	-11.61	-11.74	0.125	1.375	3.250	4.750	9.000
Vehicular A	-3.17	-4.07	-11.19	-13.01		0.125	1.375	2.875	4.250	
Vehicular B	-4.83	-2.39				0.000	1.250			

A.2 Link Simulation Results

A.2.1 HARQ Performance Evaluation

A.2.1.1 HARQ Efficiency and Number of Retransmissions

In this section, the following notation is used:

$$\frac{E_{c,avg}}{N_t} = \frac{E_{c,1}}{N_t} \cdot N_{avg} \cdot \frac{1}{T_{trans}} = \text{Average Ec/Nt}$$

$$\frac{E_{b,comb}}{N_t} = \frac{E_{c,avg}}{N_t} \cdot \frac{W}{R_{target}} = \text{Average Eb/Nt}$$

$$\frac{E_{c,1}}{N_t} = \frac{E_{c,dpcch}}{N_t} \cdot \left(1 + \frac{E_{c,e-dpdch}}{E_{c,dpcch}}\right) = \text{Single transmission Ec/Nt including DPCCH overhead}$$

$$N_{avg} = \text{Average number of transmissions needed for successful transmission} \leq T_{trans}$$

$$T_{trans} = \text{Target (maximum) number of transmissions} = 1 \text{ or } 2 \text{ or } 4$$

$$R_{target} = \text{Target data rate}$$

$$W = \text{Chip rate} = 3.84\ \text{Mcps}$$

The simulation assumptions are shown in Table A.2.1.1.1.

Parameter	Value
DPCCH Slot format	0
Channel estmation	Realistic
Inner Loop PC	Enabled
Outer Loop PC	Based on Residual BLER
PC BER	4%
PC feedback delay	1-slot
Channel	AWGN
Number of Rx antennas	2
HARQ Xrv	{0}, {0,3}, {0,3,5,7} for 1/2/4 transmissions

Table A.2.1.1.1 Simulation Assumptions

Figures A.2.1.1.1 to A.2.1.1.4 show the simulation results.

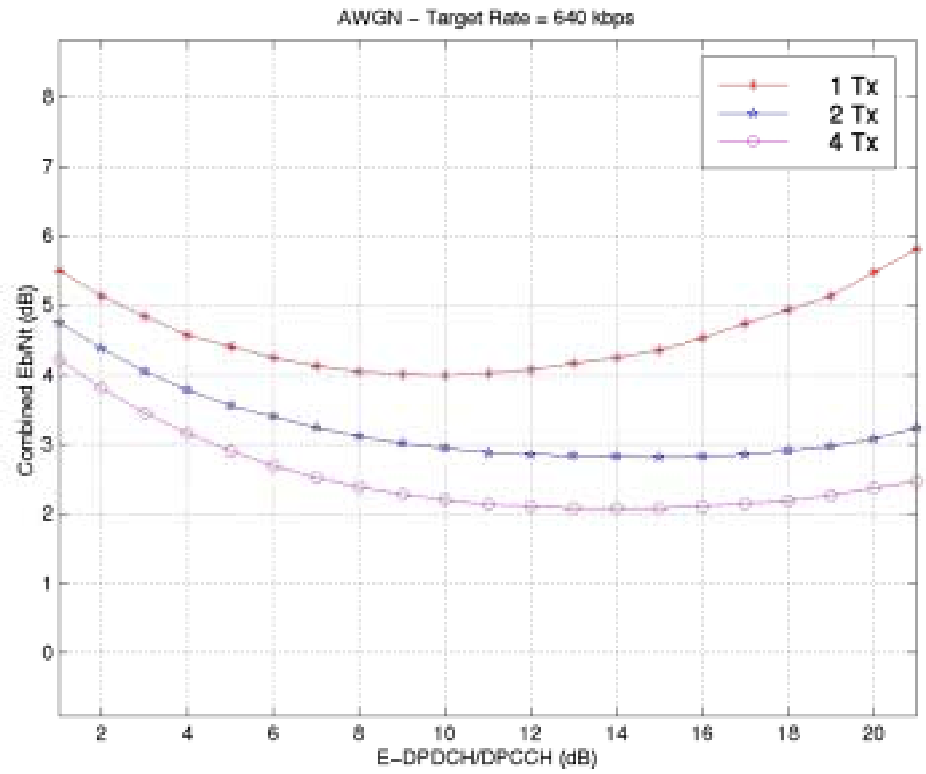

Figure A.2.1.1.1 Eb/Nt vs. E-DPDCH/DPCCH – Target Data Rate = 640 kbps

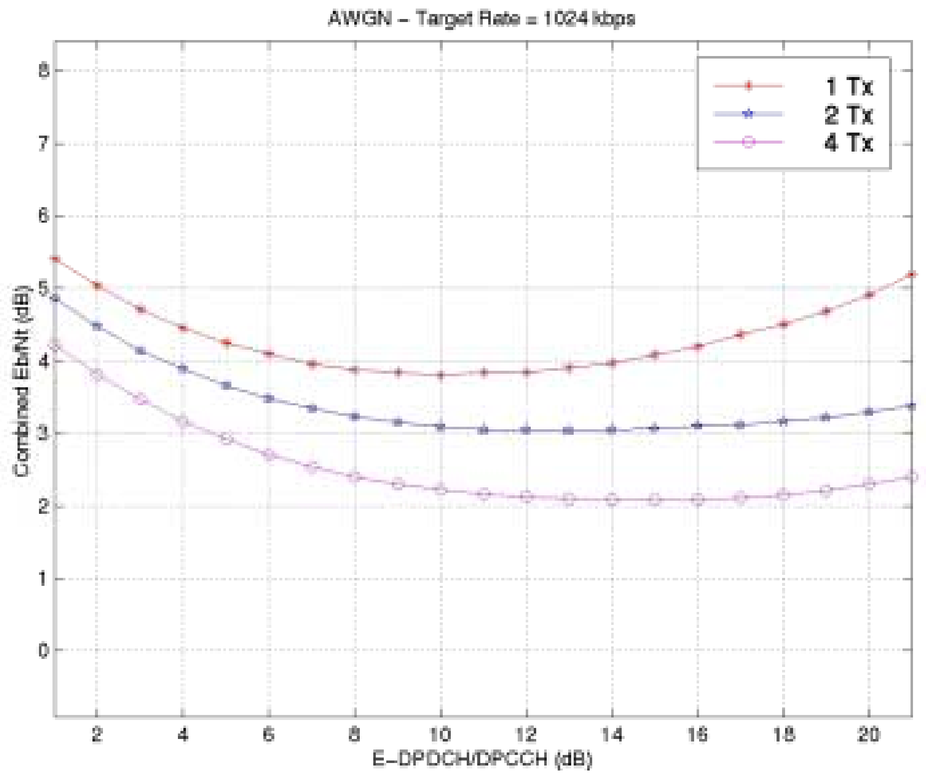

Figure A.2.1.1.2 Eb/Nt vs. E-DPDCH/DPCCH – Target Data Rate = 1024 kbps

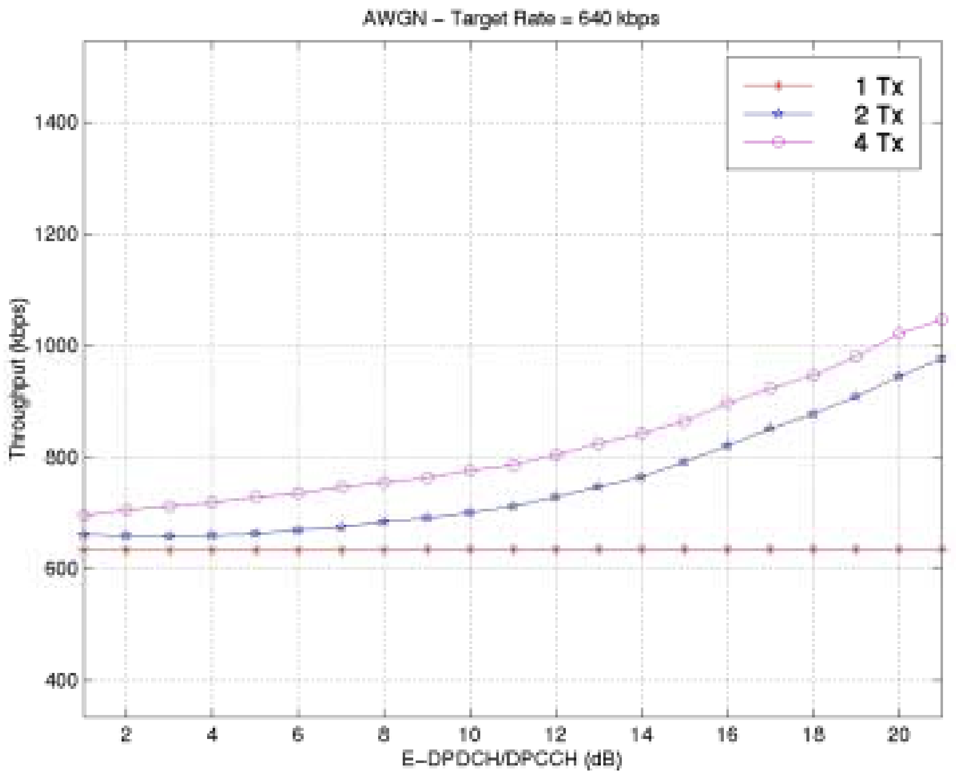

Figure A.2.1.1.3 Throughput vs. E-DPDCH/DPCCH – Target Data Rate = 640 kbps

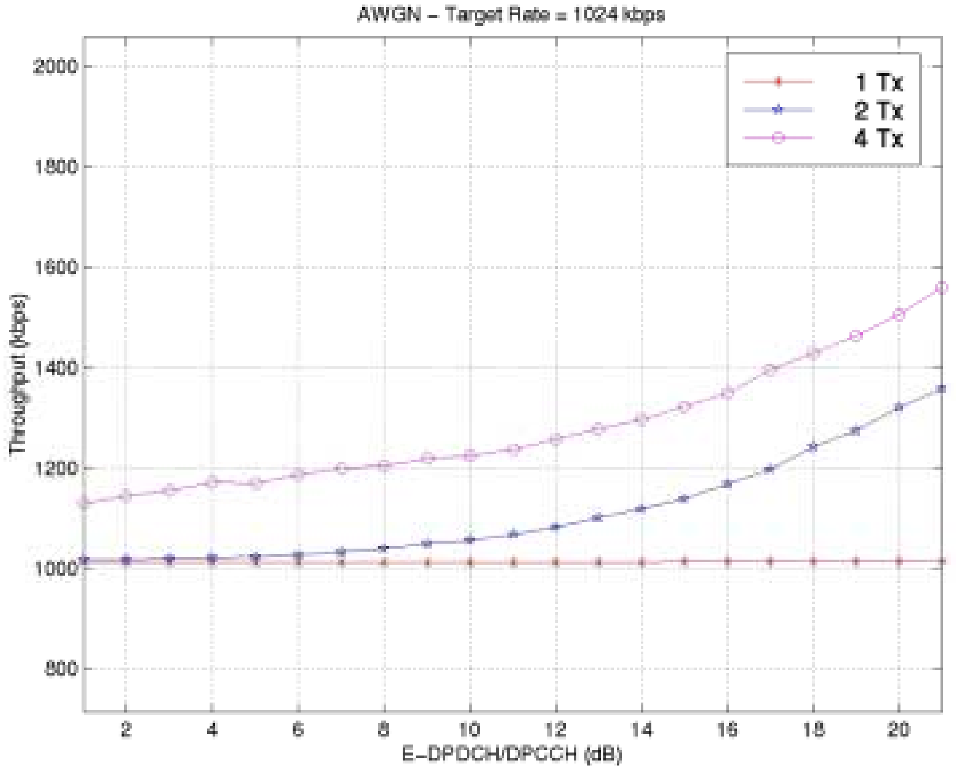

Figure A.2.1.1.4 Throughput vs. E-DPDCH/DPCCH – Target Data Rate = 1024 kbps

The optimal operating points for all cases are shown in Table A.2.1.1.2.

Target Data Rate (kbps)	MCS	Optimal EDPDCH/ DPCCH (dB)	Optimal DPCCH SNR (dB)	Optimal Eb/Nt (dB)	BLER				Average number of transmissions
					1 Tx	2 Tx	3 Tx	4 Tx	
640	4	10	-14.2	4.0	0.01	-	-	-	1.00
640	9	15	-19.1	2.8	0.60	0.01	-	-	1.60
640	19	15	-19.5	2.1	0.99	0.76	0.18	0.01	2.93
1024	7	10	-12.4	3.8	0.01	-	-	-	1.00
1024	15	13	-15.6	3.0	0.84	0.01	-	-	1.84
1024	31	15	-17.6	2.1	0.99	0.84	0.22	0.01	3.07

Table A.2.1.1.2 Optimal Operating Point – 1, 2, 4 transmissions

A.3 System Simulation Assumptions

As system level simulation tools and platforms differ between companies very detailed specification of common simulation assumptions is not feasible. Yet, basic simulation assumptions and parameters should be harmonized as proposed in the subsequent chapters. Various kinds of system performance evaluation methods may be used.

A.3.1 System Level Simulation Modelling and Parameters

A.3.1.1 Antenna Pattern

The antenna pattern [2] used for each sector, uplink and forward Link, is plotted in Figure A - 1 and is specified by

$$A(\theta) = -\min\left[12\left(\frac{\theta}{\theta_{3dB}}\right)^2,\ A_m\right] \quad \text{where} \quad -180 \le \theta \le 180$$

, θ_{3dB} is the 3dB beam width, and $A_m = 20dB$ is the maximum attenuation.

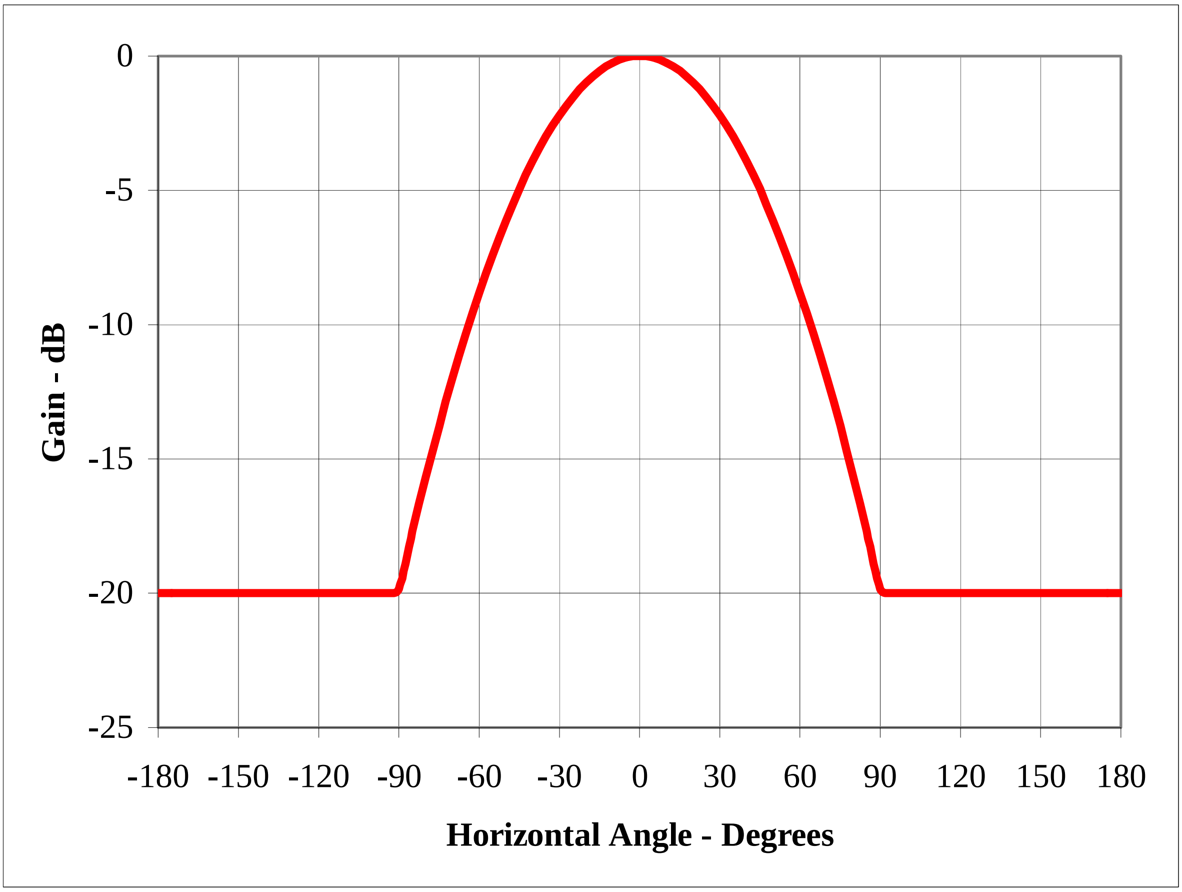

Figure A - 1 - Antenna Pattern for 3-Sector Cells

A.3.1.2 System Level Parameters

Table A - 7 below shows the general system level parameters, to be used both in the reference case, and in the new schemes proposed for Enhanced Uplink DCH. Table A - 8 shows the system level parameters to be used in the reference case.

Table A - 7 – General System Level Simulation Parameters

Parameter	**Explanation/Assumption**	**Comments**
Cellular layout	Hexagonal grid, 3-sector sites	
Site to Site distance	2800 m 1000 m	
Antenna pattern	0 degree horizontal azimuth is East 70 degree (-3dB), 20dB front-to-back ratio	Only horizontal pattern specified See Section 3.1.1.
Propagation model	$L = 128.1 + 37.6\ Log_{10}(R)$ (see [6])	R in kilometres
Downlink CPICH power	-10 dB	Relative to the maximum power
Other downlink common channels	-10 dB	Relative to the maximum power
Slow fading	Similar to UMTS 30.03, B 1.4.1.4	
Std. deviation of slow fading	8.0 dB	Log-Normal Shadowing
Correlation between sectors	1.0	
Correlation between sites	0.5	See Annex B
Correlation distance of slow fading	50 m	See D,4 in UMTS 30.03.
Carrier frequency	2000 MHz	
Node B antenna gain plus Cable Loss	14 dBi	
Node B RX diversity	Uncorrelated 2-antenna RX diversity	Maximal ratio combining
UE antenna gain	0 dBi	
Maximum UE EIRP	21 dBm	Also 24 dBm can be simulated additionally, however 21 dBm should be the main case.
BS total Tx power	43 dBm	
Active set size	Up to 3	Maximum size
Uplink system noise	–102.9 dBm	
Specify Fast Fading model	Jakes spectrum where Doppler based on speed.	Generated e.g. by Jakes or by Filter approach
Soft Handover Parameters	Window_add = 4 dB, Window_drop = 6 dB	Window_add: The signal from a BS has to be at highest this amount smaller than the current active set's best BS's signal for a BS to be added in the active set. Window_drop: When the signal from a BS has dropped below the active set's best BS's signal minus this parameter, the BS will be dropped from the active set.

Uplink Power Control	Closed-loop power control delay: one slot	Power control feedback: BER = 4% for a Node-B - UE pair.
Short term average Rise over Thermal (Uplink Received Power Normalized by Thermal Noise Level)	x dB [3]	The percentage of time the short term average rise over thermal is above the x dB target should not exceed 1%. Short term average Rise over thermal for the default two receiving antenna mode is the result of filtering the instantaneous rise ½[(Io1+No)/No + (Io2 + No)/No] with the filter described in Annex C, where the total received signal power at antenna i is defined as Ioi, I=1,2.
Delays between network elements.	Document [7] is resource and starting point for delay information between different network elements for release 5.	

Table A - 8 - System Level Simulation parameters used in the reference rel99/rel4/rel5 case

Parameter	Explanation/Assumption	Comments
Method included in the reference case	Rel'99 / Rel'4 / Rel'5 System with TFC selection	The parameters defined based on Rel'99 / Rel'4 / Rel'5 specifications
User data rates in TFCS allocated to the UE	TFCS1: 8, 16, 32, 64, 128, 256, 384 kbit/s TFCS2: 8, 16, 32, 64, 128, 256, 384, 768, 1000 kbit/s	One of these two TFCS is to be included in the TFC selection modelling.
TTI	10 ms	Maximum TTI in the TFC
Ptx estimation error in TFC selection	The error is within ±2 dB with 90% certainty.	Error is Log normally distributed around zero mean with std = 1.2159 dB.
Delay for moving TFC into blocked state in TFC selection 9.33 ms + T_{notify} + T_{modify}+ T_{L1_proc} + T_{align_TTI}	60 ms	As defined in current specification, assuming max TTI in the TFC being T_{TTI} =10 ms and no codec which needs rate adjustment.
Delay for moving TFC back into supported state in TFC selection 9.33 ms + T_{notify} + T_{modify}+ T_{L1_proc} + T_{align_TTI}	60 ms	As defined in current specification, assuming max TTI in the TFC being T_{TTI} =10 ms and no codec which needs rate adjustment.

In the proposed schemes for Enhanced Uplink DCH, following parameters are defined in more detail:

- TFC selection method should be used with the same parameters as in the reference case, if there is no clear reason why it does not fit to the scheme.
- Used data rates and transport formats
- Parameters and other details of scheduling

A.3.1.3 Signaling Errors

Signaling errors may be modeled and specified as the examples in Table A - 9.

Table A - 9 - WCDMA Signaling Errors

[3] Note that the final value for the rise outage threshold and its exact use will be determined later as simulation and analytical results are generated by proponent companies. One reason for having a rise outage threshold is to guarantee acceptable voice call quality and reliable signaling given autonomously or explicitly scheduled data UEs on the Release 99/4/5 or enhanced uplink channel.

Signaling Channel	Errors	Impact
ACK/NACK channel	Misinterpretation, misdetection, or false detection of the ACK/NACK message	Transmission (frame or encoder packet) error or duplicate transmission
Scheduling related signaling	Misinterpretation of feedback information	Potential transmission errors

For H-ARQ, if an ACK is misinterpreted as a NACK (duplicate transmission), the packet call throughput should be scaled down by (1-p_{ACK}), where p_{ACK} is the ACK error probability. Otherwise the signaling errors will be explicitly modeled to properly account for them.

A.3.1.4 Downlink Modeling in Uplink System Simulation

In addition to modelling CPICH transmission for the purpose of active set selection, only feedback errors for e.g. power control, acknowledgements, scheduling related signaling etc. need to be modeled. Thus explicit modeling of the downlink channels is not required.

A.3.2 Uplink measurement accuracy

Measurement errors for taking instantaneous (e.g. 0.667 ms) samples of Received total wideband power (RTWP), (also called Io), can be modeled as a lognormal process with standard deviation and mean as given below and in keeping with RTWP requirements given in specification 25.133 [8] (see specifically section 9.2 and Annex A.9 in 25.133).

Absolute interference rise error mean: 0

Absolute interference rise error std. dev.: 4 / 1.28

Relative interference rise error mean: 0

Relative interference rise error std dev.: 0.5 / 1.28

A.3.2.1 Uplink power control

Inner loop power control update rate is assumed to be 1500Hz in keeping with release 5. Inner loop power control is applied to all uplink channels including the EUDCH, the proponent should indicate otherwise.

Outer loop power control is needed so that the DPCCH can meet minimum required Ec/Nt. Outer loop power control can be active at all times by using a Rel-99 Zero-block CRC DPDCH which will also keep the DPCCH at the minimum required received Ec/Nt for demodulation of the EUDCH and other uplink control channels

A.3.3 System Simulation Outputs and Performance Metrics

A.3.3.1 Output metrics for data services

The following statistics related to data traffics should be generated and included in the evaluation report for each scheme. If wrap-around is used [9], statistics are collected from all cells, otherwise at least from “center cell(s)”. If wrap-around is not used statistic collection is taken from “center cell(s)” and at least two tiers of cells around the “center cell” site. A frame as used below is also referred to as a transport block and consists of information bits, CRC, and tail bits.

1. **Average cell throughput [kbps/cell]** is used to study the network throughput performance, and is measured as

$$R = \frac{b}{k \cdot T},$$

where b is the total number of correctly received data bits in the uplink from all data UEs in the simulated system over the whole simulated time, k is the number of cells in the simulation and T is the simulated time. In the case of only evaluating the center cell site, k is the number of sectors.

2. **Average packet call throughput [kbps]** for user i is defined as

$$R_{pktcall}(i) = \frac{1}{K}\sum_{k=1}^{K}\frac{\text{good bits in packet call k}}{(t_{end\_k} - t_{arrival\_k})}$$

where k = denotes the k^{th} packet call from a group of K packet calls where the K packet calls can be for a given user i , $t_{arrival\_k}$ = first packet of packet call k arrives in queue, and t_{end_k} = last packet of packet k is received by the Node-B. Note for uncompleted packet calls, t_{end_k} is set to simulation end time. The mean, standard deviation, and distribution of this statistic is to be provided.

3. **The packet service session FER** is calculated for all the packet service sessions. A packet service session FER is defined as the ratio

$$FER_{session} = \frac{n_{erroneous\_frames}}{n_{frames}},$$

where $n_{erroneous\_frames}$ is the total number of erroneous frames in the packet service session and n_{frames} is the total number of frames in the packet service session. These individual packet service session FERs from all packet service sessions (from all UEs) form the distribution for this statistic. The mean, standard deviation, and the distribution of this statistic is to be provided.

A Definition of a Packet Service Session: A Packet Service Session contains one or several packet calls depending on the application. Packet service session starts when the transmission of the first packet of the first packet call of a given service begins and ends when the last packet of the last packet call of that service has been transmitted. (One packet call contains one or several packets.) Note, that FER statistics are only collected from those frames during which UE is transmitting data.

4. **The residual FER** is calculated for each user for each packet service session. A packet service session residual FER is defined by the ratio

$$FER_{residual} = \frac{n_{dropped\_frames}}{n_{frames}},$$

where $n_{dropped\_frames}$ is the total number of dropped frames in the packet service session and n_{frames} is the total number of frames in the packet service session. A dropped frame is one in which the maximum ARQ or HARQ re-transmissions have been exhausted without the frame being successfully decoded. In the case of HARQ the proponent should indicate whether he is including RLC initiated re-transmissions or not. The mean, standard deviation, and distribution of this statistic over all the packet service sessions in the simulation is to be provided.

5. **The averaged packet delay per sector** is defined as the ratio of the accumulated delay for all packets for all users received by the sector and the total number of packets. The delay for an individual packet is defined as the time between when the packet enters the queue at transmitter and the time when the packet is received successively by the base station. If a packet is not successfully delivered by the end of a run, its ending time is the end of the run.

6. **The histogram of averaged packet delay per user**. The averaged packet delay is defined as the ratio of the accumulated delay for all packets for the user and the total number of packets for the user. The delay for a packet is defined as in 2.

7. **The scattering plot of data throughput per user vs. its averaged packet delay**. The data throughput and averaged packet delay per user are defined as in 3 and 2, respectively**.**

8. **The uplink TxP** is the ideal measured UE TxP at the UE antenna connector. This is collected from all the UEs at desired intervals. A distribution of these over the simulation time is to be provided.

9. **The noise rise** is defined as the ratio of the total received wideband power and the thermal noise. Noise rise samples are taken every 0.667ms. Mean, std and the 95^{th} percentile of this and the distribution is to be provided.

A.3.3.2 Mixed Voice and Data Services

In order to fully evaluate the performance of a proposal with mixed data and voice services, simulations are repeated with different loads of voice users. The following outputs may be generated and included in the evaluation report.

1. The following cases can be simulated: no voice users (i.e., data only), voice users only (i.e., number of voice users equal to voice capacity), and $\lfloor 0.25N_{max} \rfloor$ or $\lfloor 0.5N_{max} \rfloor$ voice users with data users, where N_{max} is the voice capacity.

2. For each of the above case, all corresponding output metrics defined previously are generated, whenever it is applicable.

In addition, the following output may also be generated and included in the evaluation report:

1. A curve of sector data throughput vs. the number of voice users is generated, where the sector data throughput is defined as above.

A.3.3.3 Voice Services and Related Output Metrics

The following statistics related to voice traffics can be generated and included in the evaluation report.

1. **Voice capacity**. Voice capacity is defined as the maximum number of voice users that the system can support within a sector with certain maximum outage probability. The details on how to determine the voice capacity of a sector are described in Annex D.

2. **Percentage of blocked voice user**

A.3.3.3.1 Voice Model

An example speech (voice) model is specified in Annex D.

A.3.3.4 Packet Scheduler

The voice users' (if simulated together with the data users) transmissions are not scheduled. The data users can be scheduled or allowed to transmit in a random fashion. The exact procedure and its delay and reliability with which a UE gains the right to transmit is to be specified in detail.

A.4 System Simulation Results

A.4.1 Release-99 Performance

A.4.1.1 Release-99 Performance With Full Buffer

A.4.1.1.1 System Setup

The system performance is obtained under the following assumptions:

- Link-level curves used are generated based on the following parameters, where each pair represents (TFC,DPDCH/DPCCH): (8 kbps, 0 dB), (16 kbps, 2 dB), (32 kbps, 4 dB), (64 kbps, 7 dB), (128 kbps, 10 dB), (256 kbps, 13 dB), (384 kbps, 15 dB),
- Maximum data rate is 384 kbps
- 19 Node-B, 3-cell wrap-around layout
- Simulation duration: 200 s
    - Additional warm-up time, during which statistic is not collected: 10 s

A.4.1.1.2 Performance Without TFC Control in AWGN

The following figures present the system performance in AWGN, without TFC control, in terms of average RoT and throughput per user. Figure A.4.1.1.2.2 represents the average RoT as a function of the number of users per cell. It can be seen that as the number of users increases, the RoT increases.

Figure A.4.1.1.2.3 shows the scatter plot of the user throughputs for 5, 10 and 15 users per cell as a function of the best downlink path loss. From this figure it can be seen that as the number of users increases, the cell coverage decreases.

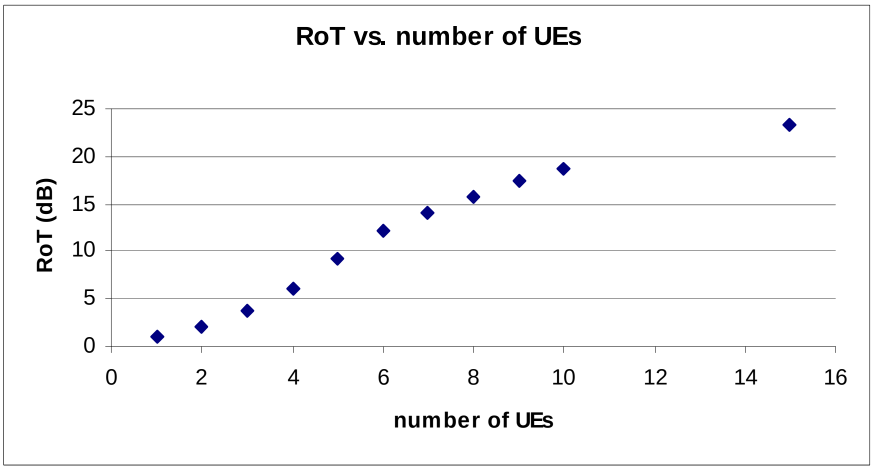

Figure A.4.1.1.2.2 Average RoT as a function of number of users per cell

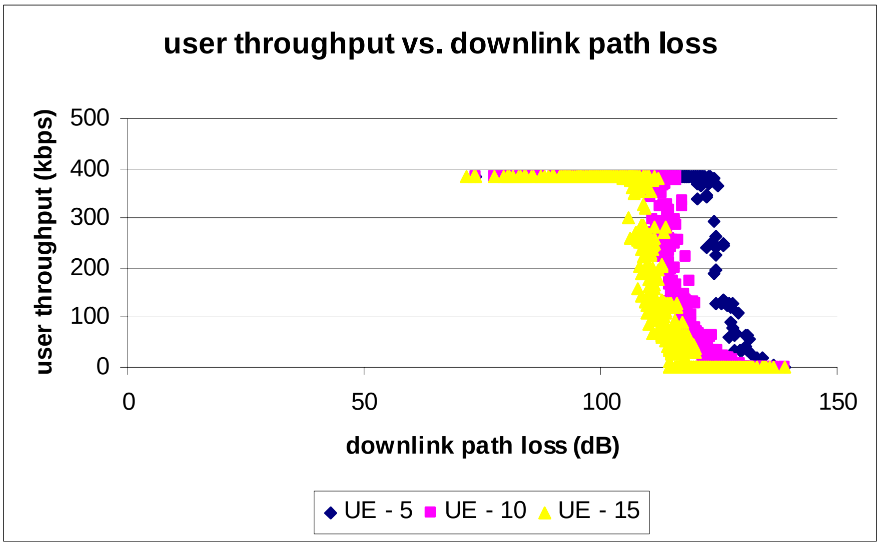

Figure A.4.1.1.2.3 Average user throughput as a function of the best downlink path loss

A.4.1.2 Release-99 Performance With Mixed Traffic Model

A.4.1.2.1 System Setup

The system performances are obtained under the following assumptions:

- Link-level curves used are generated based on the following parameters, where each pair represents (TFC,DPDCH/DPCCH): (8 kbps, 0 dB), (16 kbps, 2 dB), (32 kbps, 4 dB), (64 kbps, 7 dB), (128 kbps, 10 dB), (256 kbps, 13 dB), (384 kbps, 15 dB),
- Traffic model: FTP, Near Real Time Video, Gaming
    - The TCP parameters, as defined in Table A-13, for FTP users are: Mean(τ_2)=50 ms, Mean(τ_3)=50 ms, StdDev(τ_3)=50 ms, τ_4=0 ms if packet is in error after all physical layer retransmissions and τ_4=200 ms otherwise
- Initial FTP state is the reading time, exponentially distributed with mean of 18 s
- The Gaming traffic model parameters are as defined in the Table A-10, for Value Set 2
- Maximum data rate is 384 kbps
- 19 Node-B, 3-cell wrap-around layout
- Simulation duration: 200 s
    - Additional warm-up time, during which statistic is not collected: 10 s

A.4.1.2.2 Performance Without TFC Control in AWGN

The following figures present the system performance in AWGN, without TFC control, in terms of average RoT, throughput per user, packet call throughput per user and packet call delay.Figure A.4.1.2.2.1 represents the average RoT as a function of the number of users per cell. As the number of users increases, the RoT increases, for all traffic models.

Figure A.4.1.2.2.2 shows the scatter plot of the throughputs of the users for 10 users per cell as a function of the best downlink path loss.

Figure A.4.1.2.2.3 presents the packet call throughputs of the users in terms of the best downlink path loss, for 10 users per cell. Packet call throughput is defined as the ratio of the number of correctly received bits and the packet call delay. Packet call delay is the time between two consecutive reading periods. For Gaming users, packet call delay represents the time of a gaming session that includes the time during which the packets are generated (active period), and the time needed for transmission of the data packets accumulated during the active period. For FTP users, packet call delay is the time needed for an FTP file upload. Packet call delays are presented in Figure A.4.1.2.2.4. The packet call delay is shown for FTP and Gaming users only, since the packet call delay for Video users is not specifically defined and is actually equivalent to the simulation duration.

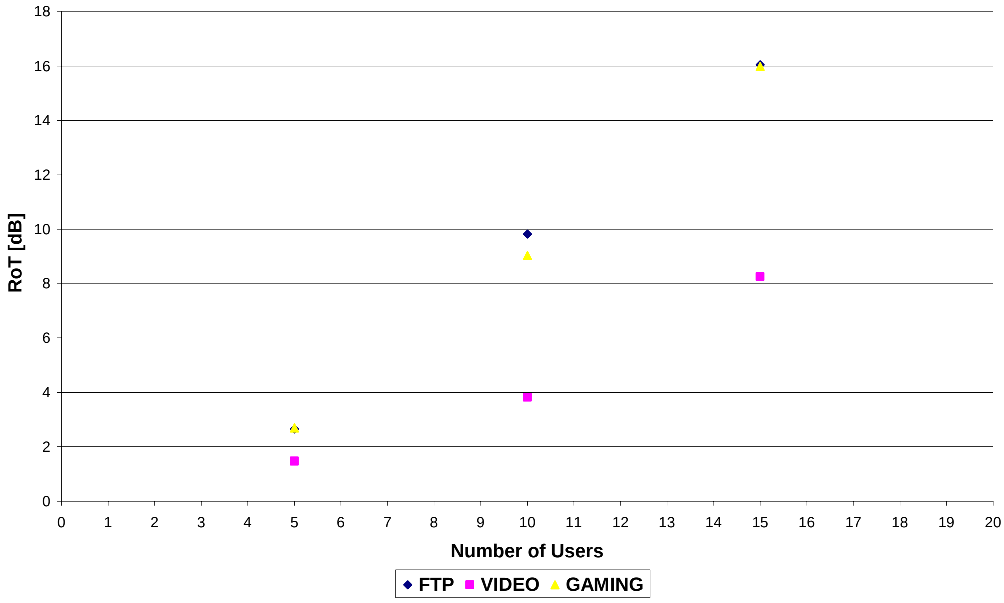

Figure A.4.1.2.2.1: Average RoT as a function of number of users per cell

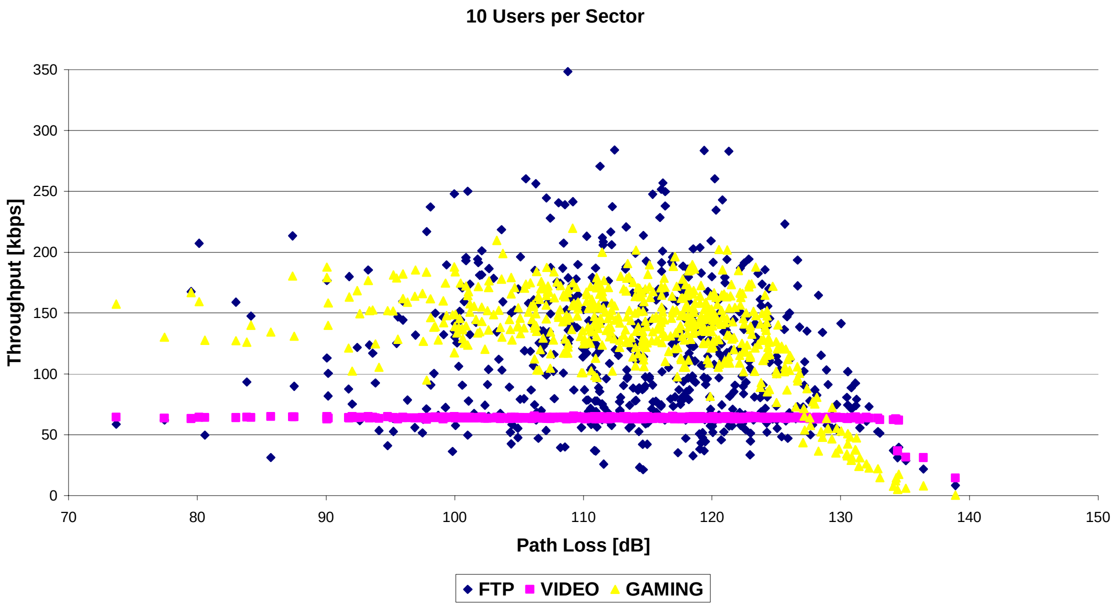

Figure A.4.1.2.2.2: Average user throughput as a function of the best link path loss for the system with 10 users per cell

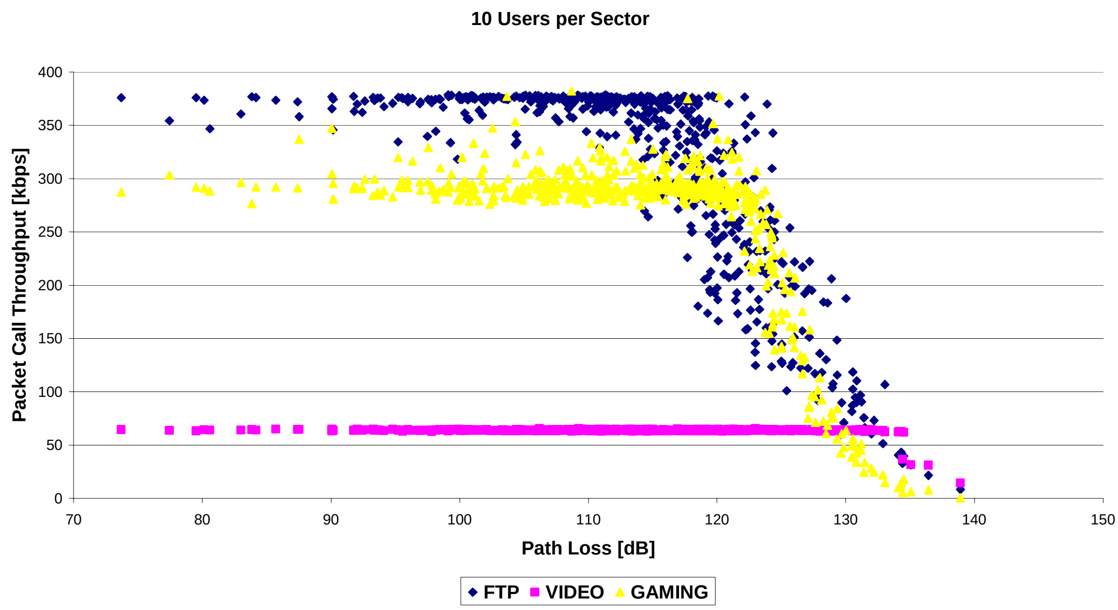

Figure A.4.1.2.2.3: Average packet call throughput as a function of the best link path loss for the system with 10 users per cell

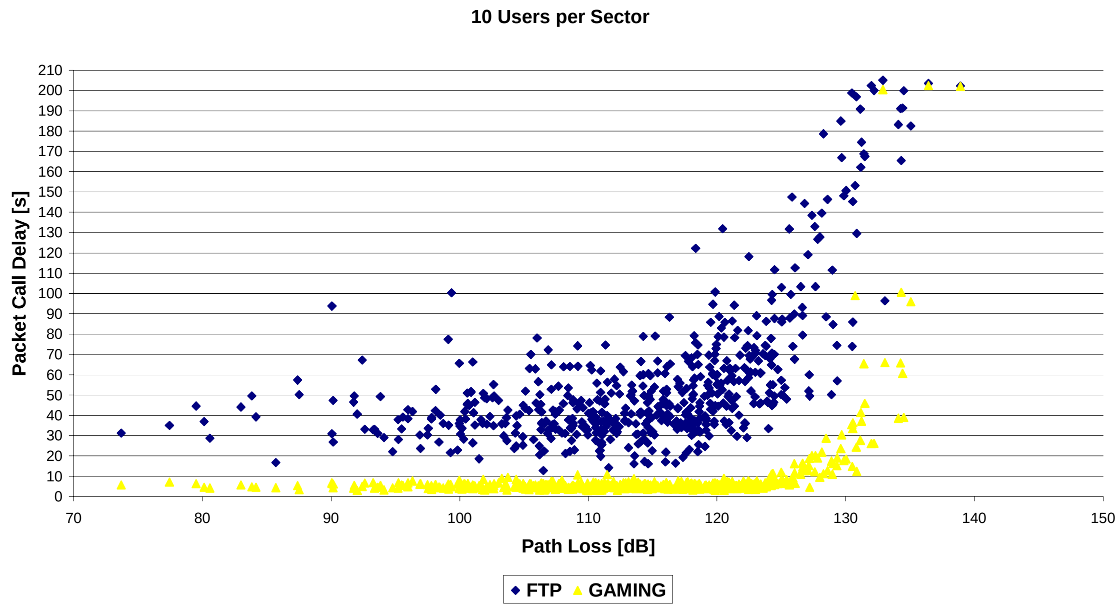

Figure A.4.1.2.2.4: Average packet call delay as a function of the best link path loss for the system with 10 users per cell

A.4.1.3 Release-99 Voice Capacity

A.4.1.3.1 System Setup

The system performance is obtained under the following assumptions:

- TTI: 20ms
- Voice rate: 12.2kbps
- DPDCH/DPCCH for each TF: (12.2 kbps, 0 dB), (SID, -4 dB), (NULL, -7 dB)

- Channel model mix: PA3 30%, PB3 30%, VA30 20% and VA120 30%
- 19 Node-B, 3-cell wrap-around layout
- Simulation duration: 500 s
    - Additional warm-up time, during which statistic is not collected: 10 s
- Rest of simulation assumptions as in Table A-7

A.4.1.3.2 Voice Capacity

Table A.4.1.3.2.1 presents the average Rot and voice outage probability.

Table A.4.1.3.2.1 Average RoT and Voice Outage Probability

Number of UEs per cell	Average RoT (dB)	Voice Outage Probability
45	2.95	0.00%
60	4.67	0.12%
75	7.54	0.47%
90	16.19	8.75%

A.5 Traffic Models

The following types data traffic models will be used in the evaluation study, a) Modified Gaming, b) near real time video and c) FTP. The traffic models are described in the following paragraph.

a) Modified Gaming Model:

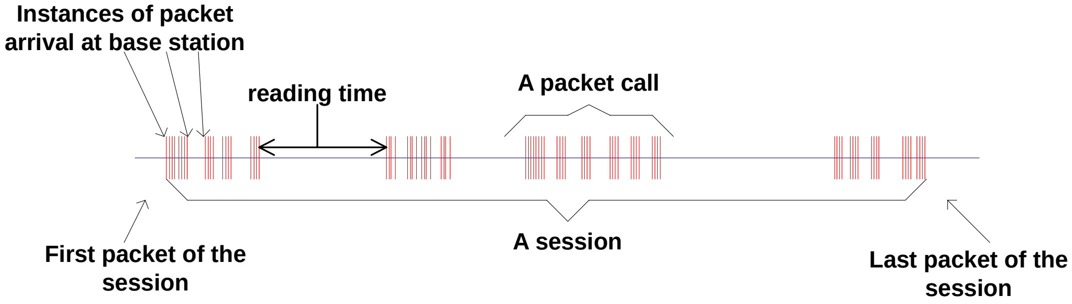

Figure A - 2 - A source packet data model with packets (datagrams) arriving as part of a packet call.

Figure A - 2 shows the source traffic model. Similar to other models it defines a packet call arrival process and within each packet call a datagram arrival process. In this model the packet session arrival process is not specified and it is assumed that packet calls are generated indefinitely (for the duration of the simulation). One may however specify a limited number of packet calls within a packet session together with an arrival probability.

For the packet call arrival process we specify the packet call (time) duration and the reading time (the time between packet calls). The reading time starts at the successful transmission of all datagrams generated during the previous packet call to emulate a closed loop transmission mode; we imagine that the application running on the UE will await acknowledgement from the network peer. Most significantly, this is a measure to ensure burstiness in the UE transmissions since it avoids excessive UE buffer accumulation, and hence continuous-like transmission, during the simulation. For the datagram arrival process we specify the packet size (bits) and the interarrival time between datagrams.

The model for this is largely derived from the so-called "Gaming" measurements [1], and therefore originally using the empirically derived distributions specified therein. However, partly as a consequence of the closed loop modeling in Figure A - 3 and for emulating future services with higher bit rates the distributions were modified slightly. For the packet call distributions, both the packet call duration and reading time have exponential distributions. The datagram size is set to a fixed value and the datagram inter-arrival distribution is a lognormal distribution. An example of the distribution is shown in Figure A - 4.

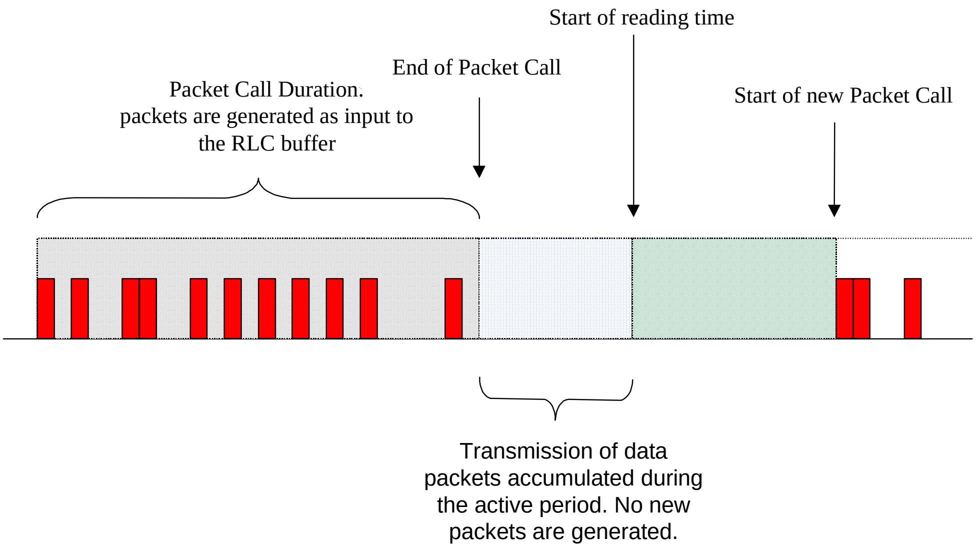

Figure A - 3 - A simple modeling approach to include closed loop transmission mode - the 'reading time' only starts after the UE RLC buffer has been emptied.

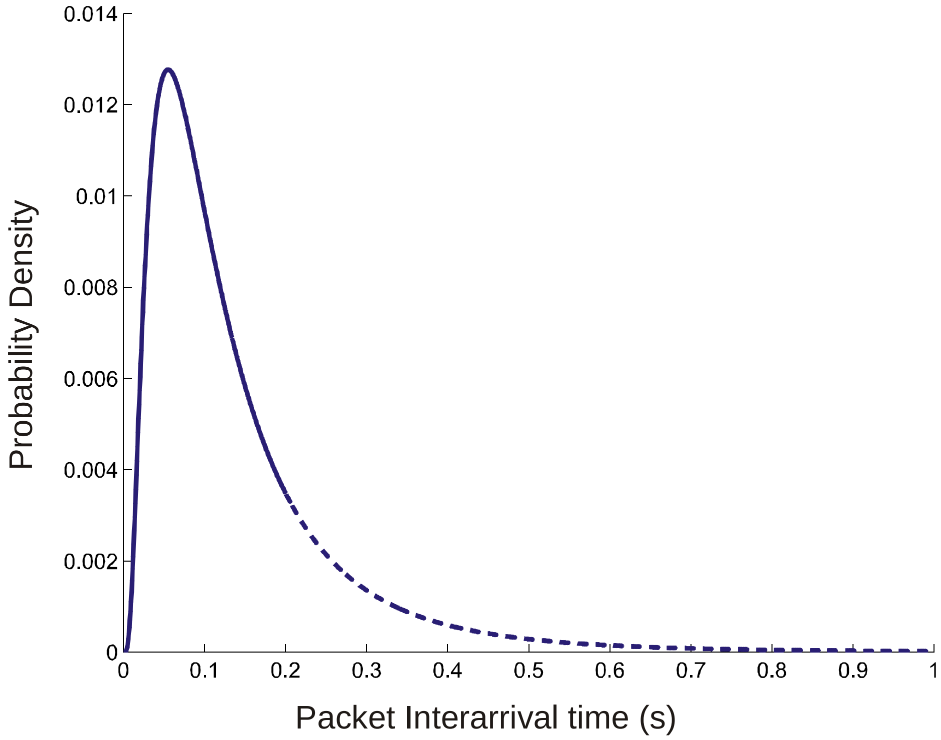

Figure A - 4 - Packet interarrival time distribution for 40 ms mean interarrival time. The packet interarrival distribution is log-normal

The model is very general and can be adjusted easily in terms of required data rates and burstiness by changing the datagram size and the mean data gram interarrival time, equivalently the mean reading time. Table A - 10 shows the parameter settings to be used in the simulations.

Table A - 10 - Parmeter Settings for the Modified Gaming model

Parameter	**Value**		**Comment**
	Value set 1	**Value Set 2**	
Mean packet call duration	5 s	5 s	Exponential distribution
Mean reading time	5 s	5 s	Exponential distribution
Datagram size	576 bytes	1500 bytes	Fixed
Mean datagram interarrival time	40 ms	40 ms	Log-normal distribution, 40 ms standard deviation
Resulting mean data rate during packet call	115 kbps	300 kbps	

The burstiness results mainly from the datagram interarrival time and the packet call reading time, while the bit rate results from the interarrival time and size of the datagrams.

b) Near Real Time Video Model:

The following section describes a model for streaming video traffic on the forward link. Figure A - 5 describes the steady state of video streaming traffic from the network as seen by the base station. Latency of starting up the call is not considered in this steady state model.

A video streaming session is defined as the entire video streaming call time, which is equal to the simulation time for this model. Each frame of video data arrives at a regular interval T determined by the number of frames per second (fps). Each frame is decomposed into a fixed number of slices, each transmitted as a single packet. The size of these packets/slices is distributed as a truncated Pareto. Encoding delay, Dc, at the video encoder introduces delay intervals between the packets of a frame. These intervals are modeled by a truncated Pareto distribution.

The parameter T_B is the length (in seconds) of the de-jitter buffer window in the Node-B used to guarantee a continuous display of video streaming data. This parameter is not relevant for generating the traffic distribution but is useful for identifying periods when the real-time constraint of this service is not met. At the beginning of the simulation, it is assumed that the Node-B's de-jitter buffer is full with (T_B x source video data rate) bits of data. Over the simulation time, data is "leaked" out of this buffer at the source video data rate and "filled" as reverse link traffic reaches the Node-B. As a performance criterion, the Node-B can record the length of time, if any, during which the de-jitter buffer runs dry. The de-jitter buffer window for the video streaming service is 5 seconds.

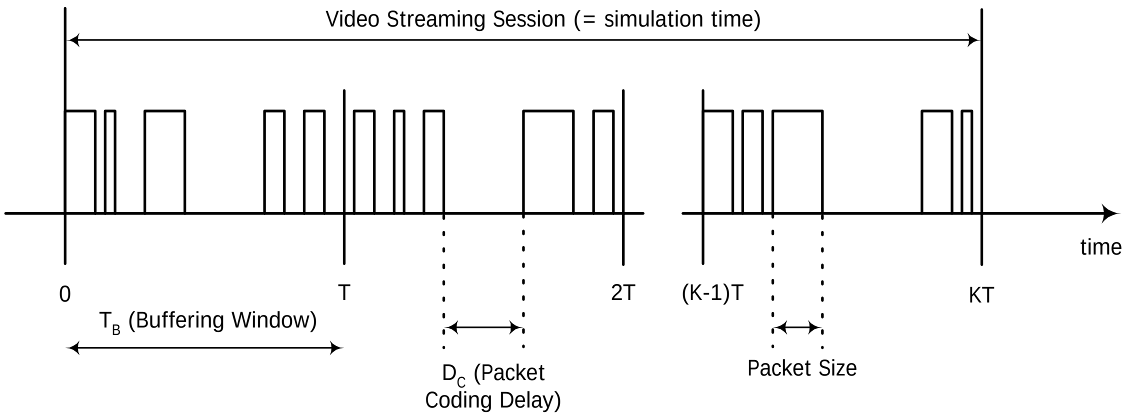

Figure A - 5 - Video Streaming Traffic Model

Using a source video rate of 64 kbps, the video traffic model parameters are defined in Table A - 11.

Table A - 11 - Typical Video Streaming Traffic Model Parameters

Information types	Inter-arrival time between the beginning of each frame	Number of packets (slices) in a frame	Packet (slice) size	Inter-arrival time between packets (slices) in a frame
Distribution	Deterministic (Based on 10fps)	Deterministic	Truncated Pareto (Mean= 100bytes, Max= 250bytes)	Truncated Pareto (Mean= 6ms, Max= 12.5ms)
Distribution Parameters	100ms	8	K = 40bytes $\alpha = 1.2$	K = 2.5ms $\alpha = 1.2$

e) FTP Model:

In FTP applications, a session consists of a sequence of file transfers, separated by *reading times*. The two main parameters of an FTP session are:

1. S : the size of a file to be transferred
2. D_{pc}: reading time, i.e., the time interval between end of download of the previous file and the user request for the next file.

The underlying transport protocol for FTP is TCP. The model of TCP connection will be used to model the FTP traffic. The packet trace of an FTP session is shown in Figure A - 6. The FTP traffic model parameters are shown in Table A - 12.

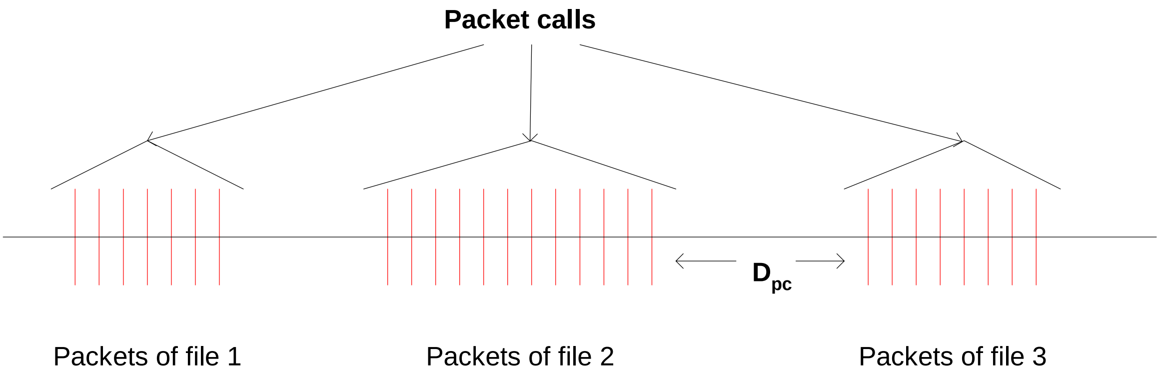

Figure A - 6 - Packet Trace in a Typical FTP Session

Table A - 12 - Typical FTP Traffic Model Parameters

Component	Distribution	Parameters	PDF
File size (S)	Truncated lognormal	Mean = 2 MB Std. Dev.= 0.722MB Max. = 5 MB	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-\ln^2(x/\mu)}{2\sigma^2}\right], x \geq 0$ $\sigma = 0.35, \mu = 14.45$
Reading time (D_{pc})	Exponential	Mean = 180 sec.	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.006$

Based on the results on packet size distribution [10], 76% of the files are transferred using and MTU of 1500 bytes and 24% of the files are transferred using an MTU of 576 bytes. For each file transfer a new TCP connection is used whose initial congestion window size is 1 segment (i.e. MTU).

The three-way handshake mechanism for TCP connection set-up and release is shown in Figure A-7.

After the call setup process is completed, the procedure for a UE to set up a TCP session is as follows:

1. UE sends a 47-byte[4] SYNC packet and wait for an ACK from remote server.
2. UE starts TCP in slow-start mode (The ACK flag is set in the first TCP segment).

The procedure for a UE to release the TCP session is as follows:

1. UE sets the FIN flag in the last TCP segment.
2. UE receives ACKs for all TCP segments from the remote server and terminates the session.

[4] The TCP/IP header of 40 bytes + 7 bytes PPP framing overhead = 47 bytes for the SYNC packet.

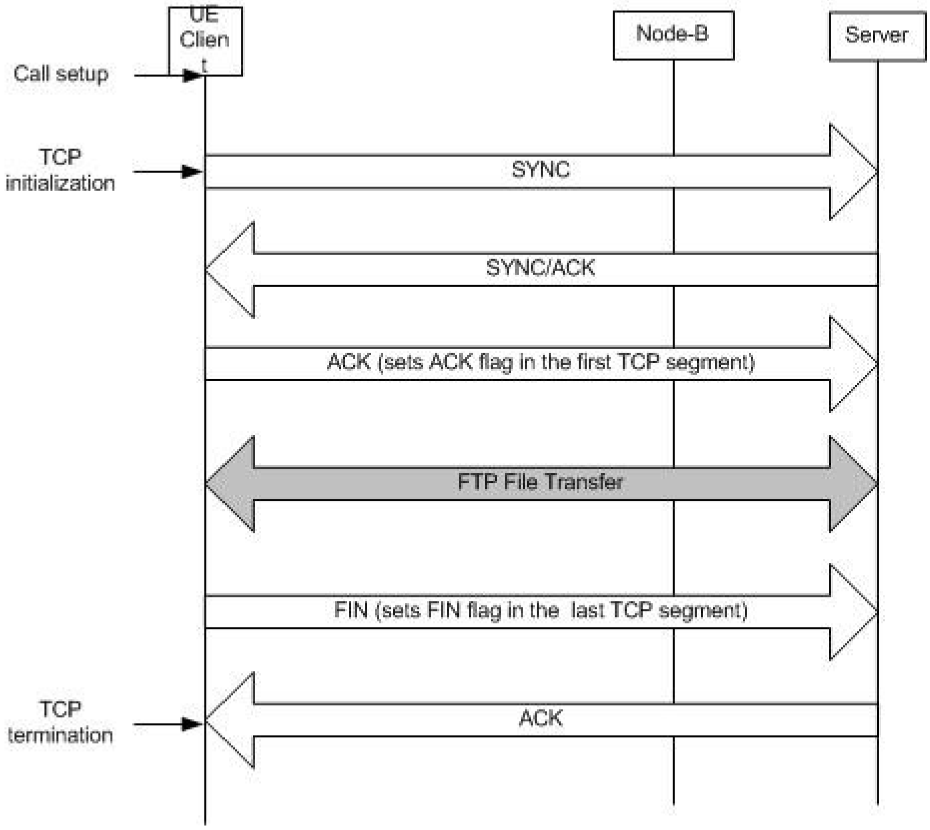

Figure A-7: Modeling of TCP three-way handshake

The amount of outstanding data that can be sent without receiving an acknowledgement (ACK) is determined by the minimum of the congestion window size of the transmitter and the receiver window size. After the connection establishment is completed, the transfer of data starts in slow-start mode with an initial congestion window size of 2 segments. The congestion window increases by one segment for each ACK packet received by the sender. This results in an exponential growth of the congestion window.

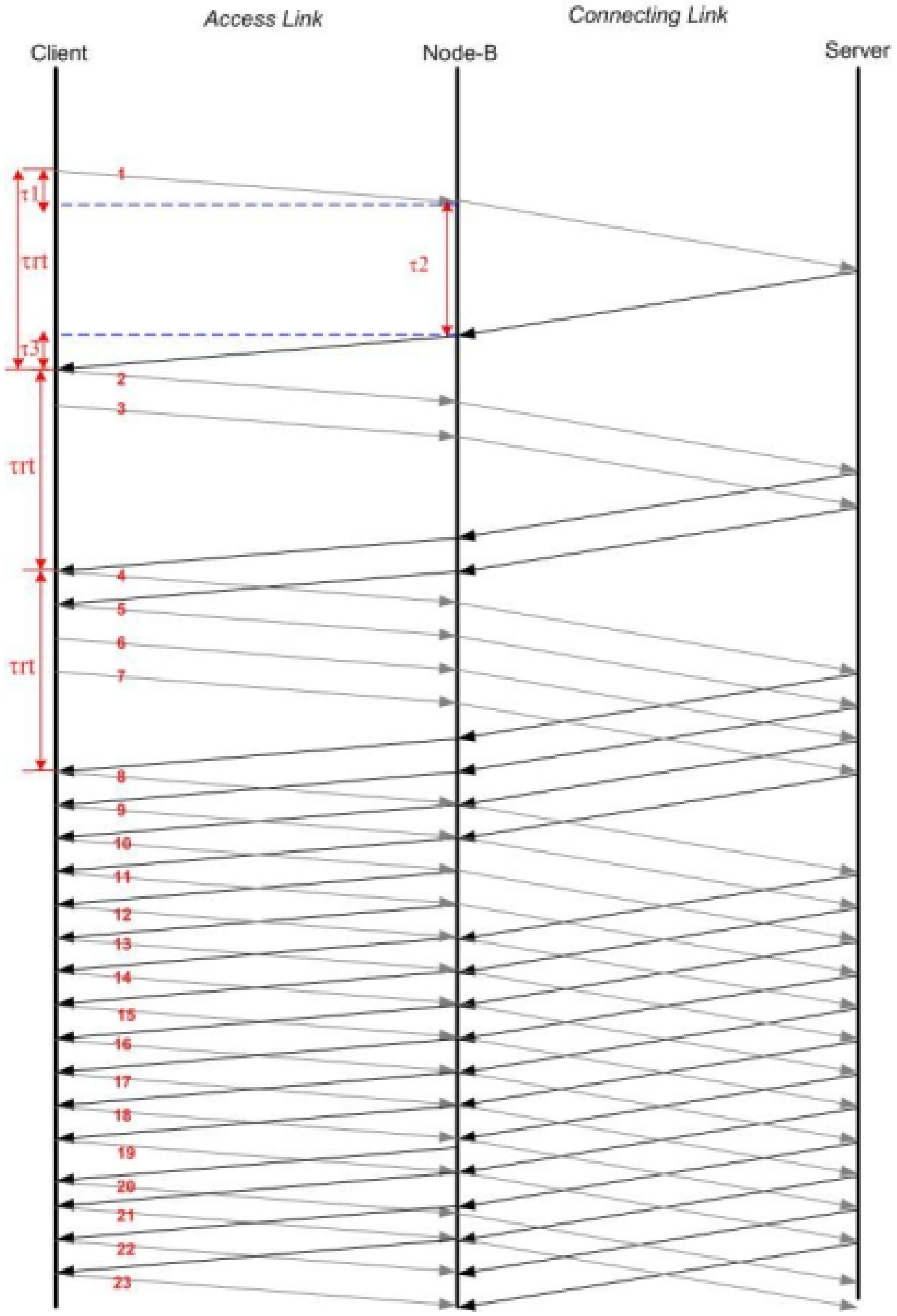

Figure A-8: TCP Flow Control During Slow-Start; τ_l = Transmission Time over the Uplink; τ_{rt} = Roundtrip Time

The round-trip time in Figure A-8, τ_{rt}, consists of two components:

$$\tau_{rt} = \tau_{cr} + \tau_l$$

where

- $\tau_{cr} = \tau_2 + \tau_3 + \tau_4$
    - τ_2 = Nominal time taken by a TCP data segment to travel from Node-B to the server plus the time taken by an ACK packet to travel from the server back to Node-B
    - τ_3 = Time taken by the ACK to travel from Node-B to client.
    - τ_4 = Constant delay to account for RLC retransmissions (nominally zero)
- τ_1 = Transmission time taken by TCP data segment from the client to Node-B

The individual delay distribution parameters are given in Table A-13.

Table A-13 **Delay components in the TCP model for the RL upload traffic**

Delay component	**Symbol**	**Value**
The uplink transmission time of a TCP data segment from the client to the Node-B	τ_1	Determined by uplink throughput
The sum of the time taken by a TCP data segment to travel from Node-B to the server and the time taken by an ACK packet to travel from the server to Node-B	τ_2	Exponential distribution Mean = x ms.
The time taken by a TCP data segment to travel from Node-B to the client.	τ_3	Lognormal distribution Mean = y1 ms Standard deviation = y2 ms
Increased delay to account for RLC retransmissions from residual uplink physical layer BLER	τ_4	Constant = 0 ms, if packet is not in error after all physical layer retransmissions = z ms, else

From Figure A-8, during the slow-start process, the UE receives two segments back-to-back after an interval of τ_{cr} for every ACK packet received.

The upload procedure is illustrated in Figure A-9 and described as follows.

1. Let S = size of the FTP upload file in bytes. Compute the number of packets in the file, N = $\lceil$S/(MTU-40)$\rceil$. W = size of the congestion window of TCP. Initially, W = 2
2. If N>W, then W packets are put into the queue for transmission; otherwise, all packets of the file are put into the queue for transmission in FIFO order. Let P = the number of packets remaining to be transmitted beside the W packets in the window. If P=0, go to step 6
3. Wait until a packet of the file in the queue is transmitted over uplink
4. Schedule arrival of next two packets (or the last packet if P=1) of the file after the packet is successfully ACKed. If P=1, then P=0, else P=P-2
5. If P>0 go to step 3
6. End.

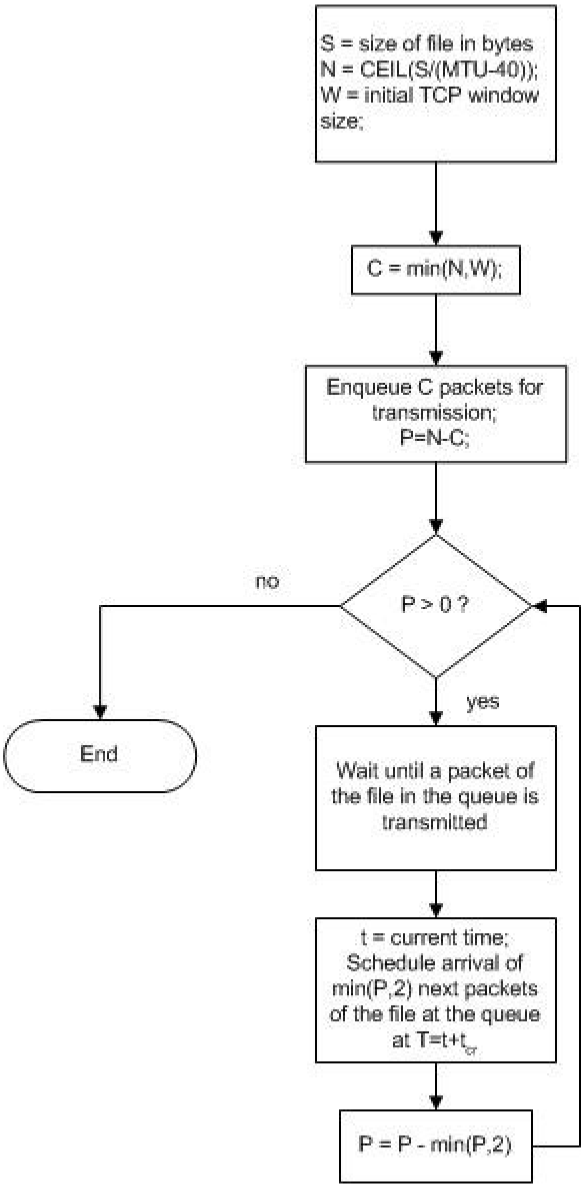

Figure A-9 Packet Arrival Process for the Upload of a File Using TCP

Annex B:
Lognormal description

The attenuation between a mobile and the ith cell site is modeled by

$$L_i = k_o D_i^{-\mu} 10^{X_i/10} R_i^2$$

where D_i is the distance between the mobile and the cell site, μ is the path loss exponent and X_i represents the shadow fading which is modeled as a Gaussian distributed random variable with zero mean and standard deviation σ. X_i may be expressed as the weighted sum of a component Z common to all cell sites and a component Z_i which is independent from one cell site to the next. Both components are assumed to be Gaussian distributed random variables with zero mean and standard deviation σ independent from each other, so that

$$X_i = aZ + bZ_i \text{ such that } a^2 + b^2 = 1$$

Typical parameters are $\sigma = 8.9$ and $a^2 = b^2 = 1/2$ for 50% correlation. The correlation is 0.5 between sectors from different cells, and 1.0 between sectors of the same cell.

Annex C: Uplink Rise Outage Filter

To determine average interference rise outage a short term average rise filter is defined.

A simple 3-tap rectangular filter is used to compute the ratio of total uplink received power to thermal noise over a radio frame interval (2 ms). The filter is applied to each set of three Rssi/thermal noise samples computed every 0.67 ms.

Z(k) = (Rssi[j]+Rssi[j+1]+Rssi[j+2])/(3*thermal noise), j=3k

where
Rssi = ½[(Io1+No)/No + (Io2 + No)/No]

No – thermal noise

Ion – uplink CDMA interference for antenna n, n=1 primary, n=2 diversity antenna.

Annex D: Speech Source (Markov) Model

The simplified speech source model with an average voice activity of 0.32 is given by

```
IF PrevState=0 then
    IF RAND()<0.01 then
            NewState=1  /* go to voice active state */
    Else
            NewState=0 /* remain in voice inactive state */
Else
    IF RAND()<0.9785 then
            NewState=1  /* remain in voice active state */
    Else
            NewState=0 /* go to voice inactive state */
```

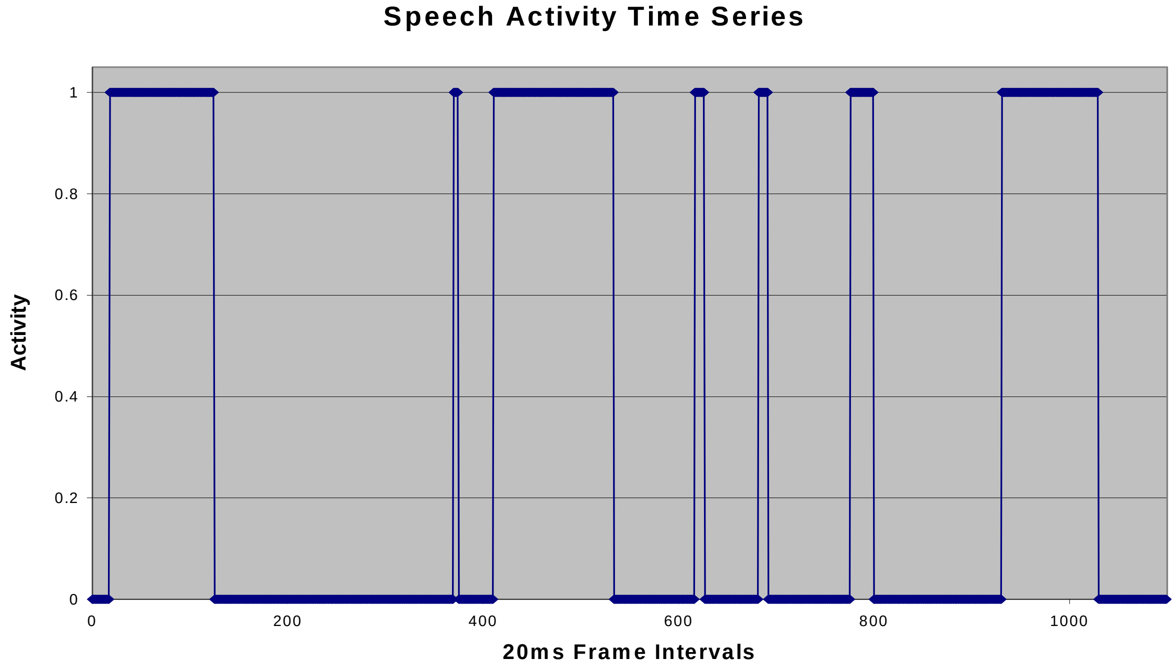

Figure D – 1 - Speech Source Example using simple Markov Model

Voice user's should meet an outage criteria which can be defined as:

a. average FER being less than 2%,

b. short term FER exceeding 2% no more than 10% of the time.

The short term FER of the voice service is calculated by averaging over 2 seconds. An AMR vocoder with a rate of 12.2 kbps will be used. The uplink voice activity factor should be set to 0.32 by randomly choosing on and off periods of appropriate duration. A simple speech source model is given above.

Annex E: Modeling of the effect of channel estimation errors on Link performance

As mentioned in Section A.1.1, the effect of channel estimation errors on link performance should be modeled for an accurate comparison of different techniques. Two methods for modeling this effect are provided in [13]. The methods described are applicable to the Quasi-static approach discussed further below. We provide below a brief overview of techniques used in [11]:

- Demodulation with imperfect channel estimates affects the SNR of the demodulated symbols. The SNR of the demodulated symbol – as seen by the turbo decoder – can be characterized *analytically*. This SNR is a function of the packet parameters such as transport block size and data rate, transmit data and pilot energies, channel gain, interference power, quality of channel estimates and combining method. Note that all of the parameters would already be generated in a system level simulation and nothing additional needs to be generated for this approach. An *effective* Eb/No for the block is then readily computed (analytically). The probability of error for the transmission is then obtained by using appropriate lookup curves (after adjusting the analytically calculated effective Eb/No by applying the Doppler penalty, puncturing penalty, and other terms, as appropriate). See [11] for more details.

In cases that do *not* involve the use of H-ARQ combining, in addition to the methods in [11], the following method may be used:

- FER Vs traffic Eb/No curves are generated for each TFC, over each fading channel model, via link level simulations. A family of curves is produced for each data rate with each curve being parameterized by the average pilot SNR over the frame. For a single packet transmission in the system simulation, the average pilot

SNR during the frame, and the received traffic channel Eb/No are computed. Performance is read off from the corresponding error curve (one which is parameterized by the same pilot SNR) obtained in the link level simulations, at the received traffic channel Eb/No value observed in the system simulation. If an error curve for this average pilot SNR does not exist for this TFC, the FER curve for this average pilot SNR is interpolated from the curves for pilot SNR immediately above and below this value, and read at the same received traffic Eb/No.

If the effect of channel estimation errors is not modeled, then several techniques, such as the ones in [3], [5] or [6], may be used:

1. Quasi-static approach [5] (QSA) with appropriate Doppler, Demapping, Puncturing penalties.

2. The modelling of link level performance at the system level is done with E_b/N_0 to BLER mapping, called the "Actual Value Interface" (AVI), described in [3].

If a comparison of schemes is based on such models – that do not incorporate the effect of channel estimation errors – then justification should be provided for not accounting for this effect.

Annex F: Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
10-2002	RAN1 #28bis	R1-02-1218			Initial TR skeleton presented for discussion		0.0.1
10-2002	RAN1 #29	R1-02-1259			Modifications to the document structure	0.0.1	0.0.2
12-2002	RAN1 #30	R1-02-1271 + comments			Requirements to chapter 5 Requirements	0.0.2	0.0.3
12-2002	RAN1 #30	R1-030065			Traffic models to Annex A	0.0.2	0.0.3
12-2002	RAN1 #30	R1-030066			Simulations assumptions to Annex A, B, C and D	0.0.2	0.0.3
01-2003	RAN1 #30	R1-030061			Reference Techniques – Uplink TFCS management by RRC signaling to chapter 6.2 and modification to Table A - 8	0.0.3	0.0.4
01-2003	RAN1 #30	R1-030062			Reference Techniques – TFC selection in UE to chapter 6.3	0.0.3	0.0.4
01-2003	RAN1 #30	R1-030126			Revised Simulations assumptions, changes to Annex A and C	0.0.3	0.0.4
01-2003	RAN1 #30	R1-030005			Added sentence to Editor's Note in chapter 8 about the physical channel timing requirements.	0.0.3	0.0.4
01-2003	RAN1 #30	R1-030131			TR25.896 version 0.0.4 agreed and promoted to 0.1.0	0.0.4	0.1.0
01-2003	RAN1 #30	R1-030150			Correction to Table A - 8 due to wrong implementation of a text proposal in Tdoc R1-030126.	0.1.0	0.1.1
02-2003	RAN1 #31	R1-030311			Revision marks approved.	0.1.1	0.2.0
02-2003	RAN1 #31	R1-030209			E-DCH definitions + additional comment	0.2.0	0.2.1
02-2003	RAN1 #31	R1-030210			Fast DCH Setup	0.2.0	0.2.1
02-2003	RAN1 #31	R1-030325			E-DCH Scheduling, 1st chapter of the text proposal added	0.2.0	0.2.1
02-2003	RAN1 #31	R1-030326			Signalling Method for Fast TFCS Restriction Control	0.2.0	0.2.1
02-2003	RAN1 #31	R1-030330			Hybrid ARQ Overview	0.2.0	0.2.1
02-2003	RAN1 #31	R1-030331			Enhanced uplink DCH physical layer structure – TTI vs HARQ structure	0.2.0	0.2.1
02-2003	RAN1 #31	R1-030332			Multiplexing Alternatives for Uplink Enhancements + additional sentence	0.2.0	0.2.1
02-2003	RAN1 #31	R1-030341			Description of Node B controlled scheduling by fast TFCS restriction control	0.2.0	0.2.1
02-2003	RAN1 #31	R1-030349			A method for Node B controlled scheduling by fast TFCS restriction control	0.2.0	0.2.1
02-2003	RAN1 #31	R1-030332			Correction to the incorrect inclusion of the text proposal	0.2.1	0.2.2
03-2003		R1-030381			Changes approved after an email review	0.2.2	0.3.0
05-2003	RAN1 #32	R1-030563			Text proposal for E-DCH scheduling in SHO	0.3.0	0.3.1
05-2003	RAN1 #32	R1-030592			Node B Controlled Time and Rate Scheduling	0.3.0	0.3.1
05-2003	RAN1 #32	R1-030594			Modifications to Section 7.1	0.3.0	0.3.1
05-2003	RAN1 #32	R1-030598			On HARQ Timing, additional row to table 8.2.1	0.3.0	0.3.1
05-2003	RAN1 #32	R1-030620			Text for TCP Modeling to Annex A.5	0.3.0	0.3.1
05-2003	RAN1 #32	R1-030621			Fast DCH Setup – Synchronization	0.3.0	0.3.1
06-2003					Editorial corrections (8.2.1 and figures A-7 and A-8)	0.3.1	0.3.2
08-2003	RAN1 #33				Addition of modified gaming model's std in table A-10	0.3.2	0.3.3
08-2003	RAN1 #33	R1-030889			Changes approved	0.3.3	0.4.0
08-2003	RAN1 #33	R1-030897			HARQ performance results with and without soft combining	0.4.0	0.4.1
08-2003	RAN1 #33	R1-030899			Changes to HARQ operation during Soft Handover	0.4.0	0.4.1

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
08-2003	RAN1 #33	R1-030911			Two new issues to be studied (Impacts of TTI to TFC selection and multiple CCTrCH to higher layers) added to chapter 8.3	0.4.0	0.4.1
08-2003	RAN1 #33	R1-030915			Shorter framesize for improved QoS overview to Chapter 7.4	0.4.0	0.4.1
09-2003	RAN1 #33	R1-030911			Added the text proposal that dropped out from the last version..	0.4.1	0.4.2
09-2003		R1-030890			R'99 voice capacity results to Annex A.4.1.3, modification of Markov model's voice outage criteria to Annex D	0.4.1	0.4.2
09-2003		R1-030891			R'99 cell throughput results with no TFC control, AWGN, full buffer added to Annex A.4.1.1	0.4.1	0.4.2
09-2003		R1-030892			R'99 cell throughput results with no TFC control, AWGN, traffic models added to Annex A.4.1.2	0.4.1	0.4.2
09-2003		R1-030896			HARQ efficiency results	0.4.1	0.4.2
09-2003					Changes approved and sent for RAN#21 for information	0.4.2	1.0.0