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Foreword

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

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

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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 TR describes the techniques behind the concept of high-speed downlink packet access (HSDPA). Furthermore, it describes how this concept should be integrated into the overall architecture of UTRA.

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.

[<seq>] <doctype> <#>[([up to and including]{yyyy[-mm]|V<a[.b[.c]]>}[onwards])]: "<Title>".

[1] 3G TS 25.123: "Example 1, using sequence field".

[2] 3G TR 29.456 (V3.1.0): "Example 2, using fixed text".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the [following] terms and definitions [given in ... and the following] apply.

Definition format

***<defined term>**: <definition>.*

example: text used to clarify abstract rules by applying them literally.

3.2 Symbols

For the purposes of the present document, the following symbols apply:

Symbol format

<symbol> <Explanation>

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

Abbreviation format

<ACRONYM> <Explanation>

4 Background and Introduction

The study item HSDPA studies enhancements that can be applied to UTRA in order to provide very high speed downlink packet access by means of a high-speed downlink shared channel (HS-DSCH).

5 Overview of Technologies considered to support UTRA High Speed Downlink Packet Access

5.1 Adaptive Modulation and Coding (AMC)

The benefits of adapting the transmission parameters in a wireless system to the changing channel conditions is well known. In fact fast power control is an example of a technique implemented to enable reliable communications while simultaneously improving system capacity. The process of modifying the transmission parameters to compensate for the variations in channel conditions is known as *link adaptation*. Another technique that falls under this category of link adaptation, is adaptive modulation and coding (AMC).

The principle of AMC is to change the modulation and coding format in accordance with variations in the channel conditions, subject to system restrictions. The channel conditions can be estimated e.g. based on feedback from the receiver In a system with AMC, users in favorable positions e.g. users close to the cell site are typically assigned higher order modulation with higher code rates (e.g. 64 QAM with R=3/4 Turbo Codes), while users in unfavorable positions e.g. users close to the cell boundary, are assigned lower order modulation with lower code rates (e.g. QPSK with R=1/2 Turbo Codes). The main benefits of AMC are, a) higher data rates are availale for users in favorable positions which in turn increases the average throughput of the cell and b) reduced interference variation due to link adaptation based on variations in the modulation/coding scheme instead of variations in transmit power.

AMC is also effective when combined with fat-pipe scheduling techniques such as those enabled by the Downlink Shared Channel. On top of the benefits attributed to fat-pipe multiplexing [2], AMC combined with time domain scheduling offers the opportunity to take advantage of short term variations in a UE's fading envelope so that a UE is always being served on a constructive fade. Figure 1 shows the Rayleigh fading envelope correlation vs. time delay for different values of Doppler frequency. The figure suggests that for lower Doppler frequencies it is possible to schedule a user on a constructive fade provided that the scheduling interval (i.e. frame size) is small and the measurement reports are timely (i.e. distributed scheduling). To take advantage of this technique, both a smaller frame size and distributed scheduling have been proposed as part of the High Speed Downlink Packet Access (HSDPA) study item.

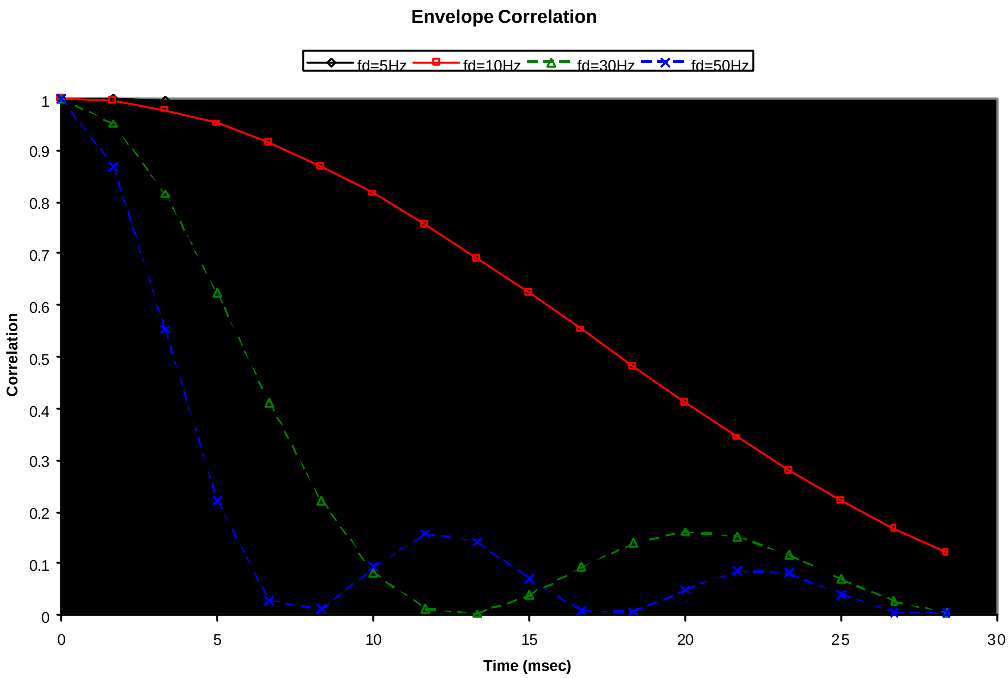


Figure 1. Envelope Correlation as a function of different doppler

5.2 Hybrid ARQ (H-ARQ)

(WG1 note: the contents of this section are for further review and alignment with WG2)

H-ARQ is an implicit link adaptation technique. Whereas, in AMC explicit C/I measurements or similar measurements are used to set the modulation and coding format, in H-ARQ, link layer acknowledgements are used for re-transmission decisions. There are many schemes for implementing H-ARQ - Chase combining, Rate compatible Punctured Turbo codes and Incremental Redundancy. Incremental redundancy or H-ARQ-type-II is another implementation of the H-ARQ technique wherein instead of sending simple repeats of the entire coded packet, additional redundant information is incrementally transmitted if the decoding fails on the first attempt.

H-ARQ-type-III also belongs to the class of incremental redundancy ARQ schemes. However, with H-ARQ-type-III, each retransmission is self-decodable which is not the case with H-ARQ-type II. Chase combining (also called H-ARQ-type-III with one redundancy version) involves the retransmission by the transmitter of the same coded data packet. The decoder at the receiver combines these multiple copies of the transmitted packet weighted by the received SNR. Diversity (time) gain is thus obtained. In the H-ARQ-type-III with multiple redundancy version different puncture bits are used in each retransmission.

AMC by itself does provide some flexibility to choose an appropriate MCS for the channel conditions based on measurements either based on UE measurement reports or network determined. However, an accurate measurement is required and there is an effect of delay. Also, an ARQ mechanism is still required. H-ARQ autonomously adapts to the instantaneous channel conditions and is insensitive to the measurement error and delay. Combining AMC with H-ARQ leads to the best of both worlds - AMC provides the coarse data rate selection, while H-ARQ provides for fine data rate adjustment based on channel conditions.

The choice of H-ARQ mechanism however is important. There are two main ARQ mechanisms - selective repeat (SR) and stop-and-wait (SAW). In SR, only erroneous blocks are re-transmitted. A sequence number is required to identify the block. Typically, in order to fully utilize the available channel capacity the SR ARQ transmitter needs to send a number of blocks while awaiting a response (or lack of it in this case). Hence when combined with H-ARQ the mobile needs to store soft samples for each partially received block. Thus mobile memory requirements can be huge. More importantly, H-ARQ requires that the receiver must know the sequence number prior to combining separate re-transmissions. The sequence number must be encoded separately from the data and must be very reliable to overcome whatever errors the channel conditions have induced in the data. Hence a strong block code is needed to encode the sequence information - increasing the bandwidth required for signaling.

5.3 Fast Cell Selection (FCS)

With Fast Cell Selection, the UE does not receive simultaneous data transmission from multiple cells and therefore performs no combining of traffic channels carrying packet data. Instead, the UE selects the best cell every frame from which it requests the data to be transmitted. The uplink DPCH is used to indicate the required cell from which the network should direct its data transmission to the UE on a frame by frame basis. This technique is a very special case of Site Selection Diversity (SSDT) and applies only to the HS-DSCH. In the case of SSDT, each cell is assigned a temporary ID and UE periodically informs a primary cell ID to the connecting cells. The non-primary cells not selected by the UE switch off their transmitter. However, in the case of Fast Cell selection, the UE selects the best cell every frame from which it wants to receive data on the HS-DSCH. HS-DSCH data is then transmitted to the UE from this cell only.

5.4 Multiple Input Multiple Output Antenna Processing

Diversity techniques based on the use of multiple downlink transmit antennas are well known; second order applications of these have been applied in the UTRA Release 99 specifications. Such techniques exploit spatial and/or polarisation decorrelations over multiple channels to achieve fading diversity gains.

Multiple input multiple output (MIMO) processing employs multiple antennas at both the base station transmitter and terminal receiver, providing several advantages over transmit diversity techniques with multiple antennas only at the transmitter and over conventional single antenna systems. If multiple antennas are available at both the transmitter and receiver, the peak throughput can be increased using a technique known as code re-use. With code re-use, each channelization/scrambling code pair allocated for HS-DSCH transmission can modulate up to M distinct data streams, where M is the number of transmit antennas. Data streams which share the same channelization/scrambling code must be distinguished based on their spatial characteristics, requiring a receiver with at least M antennas. In principle, the peak throughput with code re-use is M times the rate achievable with a single transmit antenna. Third, with code re-use, some intermediate data rates can be achieved with a combination of code re-use and smaller modulation constellations e.g. 16 QAM instaed of 64 QAM. Compared to the single antenna transmission scheme with a larger modulation constellation to achieve the same rate, the code re-use technique may have a smaller required Eb/No, resulting in overall improved system performance.

For the DSCH with conventional single antenna transmitters, a high data rate source is demultiplexed into N lower rate substreams, and the nth substream ($n = 1 \ldots N$) is spread with spreading code n (where the spreading codes indexed by $n = 1 \ldots N$ are mutually orthogonal). These substreams are summed together, scrambled and transmitted. A multiple antenna transmitter with M antennas is shown in Figure 2. It represents a typical transmitter for the multiple-input multiple-output (MIMO) antenna processing technique. The high data rate source is demultiplexed into MN substreams, and the nth group ($n = 1 \ldots N$) of M substreams is spread by the nth spreading code. The *m*th substream ($m = 1 \dots M$) of this group is transmitted over the *m*th antenna so that the substreams sharing the same code are transmitted over different antennas. These *M* substreams sharing the same code can be distinguished based on their spatial characteristics at the receiver using multiple antennas and spatial signal processing. Typically, the receiver must have at least *M* antennas to detect the signals sufficiently well; however, it is possible to perform detection using fewer than *M* antennas if more sophisticated detection algorithms are used.

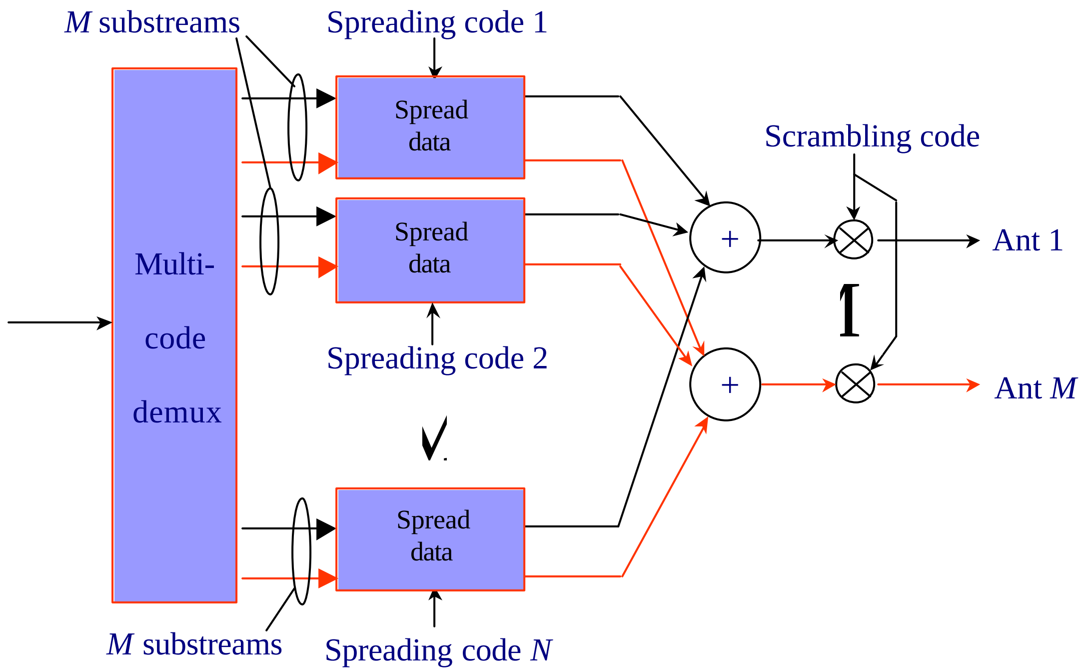


Figure 2. MIMO transmitter with M antennas

6 Proposed Physical Layer structure of High Speed Downlink Packet Access

6.1 Basic Physical Structure <frame length, update rates spreading codes, etc>

On the physical layer, HSDPA transmission should be carried out on a set of downlink physical channels (codes) shared by users at least in the time domain and possibly also in the code domain.

6.1.1 HSDPA physical-layer structure in the code domain

In the code domain, it has been proposed that HSDPA transmission would use a fixed spreading factor and multi-code transmission. The selection of such a fixed HSDPA spreading factor should be based on an evaluation of the impact on

- Performance
- UE complexity
- Flexibility (granularity in the overall allocation of capacity for HSDPA transmission)

Considerations should also be made on to what possible extent there could be any additional flexibility advantages in supporting a variable spreading factor for HSDPA, and compare these with the impact on complexity etc.

6.1.2 HSDPA physical-layer structure in the time domain

In the time domain, the support of HSDPA TTI shorter than one radio frame (10 ms) has been proposed. The length of such shorter HSDPA TTI should be selected from the set {T_{slot}, 3?T_{slot}, 5?T_{slot}, 15?T_{slot}}. The selection of such shorter HSDPA TTI should be based on an evaluation of the impact on

- Performance
- Delay
- Network and UE complexity
- Flexibility (HSDPA payload granularity)

6.2 Adaptive Modulation and Coding (AMC)

The implementation of AMC offers several challenges. First, AMC is sensitive to measurement error and delay. In order to select the appropriate modulation, the scheduler must be aware of the channel quality. Errors in the channel estimate will cause the scheduler to select the wrong data rate and either transmit at too high a power, wasting system capacity, or too low a power, raising the block error rate. Delay in reporting channel measurements also reduces the reliability of the channel quality estimate due to the constantly varying mobile channel. Furthermore changes in the interference add to the measurement errors. Hybrid ARQ (HARQ) enables the implementation of AMC by reducing the number of required MCS levels and the sensitivity to measurement error and traffic fluctuations. Finally, reducing the number of MCS levels will reduce the variance in selecting the wrong MCS level. Preliminary studies based on link and system simulation suggests that the average cell throughput is not affected if the number of MCS levels is reduced from say 7 to 3 [See Section 7.1].

There are different methods by which the HS-DSCH transport format (basically the modulation and coding scheme) can be selected:

1. Rate Request by the UE: The UE may estimate/predict the downlink channel quality and calculate a suitable HS-DSCH transport format that is reported to the Node-B as a *required transport format*. The BTS must assign UE the requested rate or have to defer its transmission.
2. Rate Determination by Node-B:

   a.)The UE may estimate/predict the downlink channel quality and calculate a suitable transport format that is reported to the Node-B as a *recommended transport format*. However, the Node-B decides on what transport format to actually use.

   b.)The UE may estimate/predict the downlink channel quality and report this to the Node-B. Based on the channel-quality report, the power control gain setting of the downlink dedicated channel, available power fraction of HS-DSCH at the Node-B etc., the network decides on what transport format to use. It may be noted that the period of the rate/channel-quality reports could be longer than the HSDPA TTI.

Conclusions:

It is too early to conclude on the details of the HS-DSCH transport-format selection and no decision is needed for the RAN1 TR.

6.3 Hybrid ARQ (H-ARQ)

6.4 Fast Cell Selection (FCS)

Fast Cell Selection implies that the UE should decide on the "best" cell for HSDPA transmission and signal this to the network.
Fast Cell Selection is conceptually similar to Site Selection Diversity Transmission (SSDT) included in Release 99. Thus FCS could inherit a significant part of the required physical-layer functionality from SSDT.

6.4.1 Physical-layer measurements for cell selection in case of fast cell selection

In case of SSDT, the "best" cell (the so-called "primary cell") is selected based on measurements of CPICH RSCP for the cells in the active set. The same measurements can be used as a basis for fast cell selection. It should be noted that other factors than measured CPICH RSCP might also affect the fast cell selection. As an example the transmit-power offset between HS-DSCH and CPICH may be different for different cells in the active set and knowledge of this offset may be useful for fast cell selection. Similarly, knowledge of the relative load of different cells may useful for fast cell selection[1]. However, this will not directly effect the physical layer and is thus not covered here.

6.4.2 Physical-layer signalling for cell selection in case of fast cell selection

In case of SSDT, the best cell is reported to the network using physical-layer (DPCCH) signalling with a maximum rate of 500 Hz. Basically identical signalling could be used for fast cell selection.
It remains to be decided if there may be requirements that a UE may have to support simultaneous independent signalling for SSDT and FCS (there may be a situation where a UE should receive HS-DSCH and dedicated transport channels on DPCH simultaneously). If this is the case there is a need for new uplink DPCCH slot formats to allow for simultaneous SSDT and FCS signalling. If this is not the case, FCS can reuse the DPCCH SSDT-signalling field and no new UL DPCCH slot formats are needed. Also note that, regardless of FCS, new UL DPCCH slot formats are most likely needed so support other HSDPA-related uplink signalling.
SSDT signalling is robust in the sense that downlink transmission from a cell is "turned-off" if and only if the SSDT signalling indicates, with sufficiently high reliability, that the cell has not been selected as primary cell. In a similar way, the FCS signalling should be robust in the sense that a cell would schedule HS-DSCH data to a UE only if the FCS signalling indicates, with sufficiently high reliability, that the cell has been selected.

6.4.3 Physical-layer signalling for transmission-state synchronisation in case of inter-Node-B FCS

If scheduling for HSDPA and termination of fast Hybrid ARQ is done at Node B there may be a need for explicit means to synchronise e.g. the scheduling and fast Hybrid ARQ states of the two Node B in case of fast inter-Node-B cell selection. One alternative is that such transmission-state synchronisation is achieved over-the-air by means of physical-layer (UL DPCCH signalling). However, if fast cell selection can select an arbitrary Node B in the active set, it is required that this uplink physical-layer signalling can be reliably detected by an arbitrary Node B in the active set. This is in contradiction to the ordinary uplink power-control strategy that ensures that uplink transmission can be reliably detected by at least one Node B in the active set but does not guarantee that uplink transmission can be reliably detected by an arbitrary Node B in the active set
At least two possible solutions can be identified:

- Use a modified uplink power-control strategy, where the UE transmit power is increased if *any* Node B in the active set requests an increase in the UE transmit power
- Use the normal uplink power-control strategy, but add a sufficiently large power offset to ensure that the transmission-state information is correctly detected with sufficiently high probability by the new Node B.

Of these alternatives the second solution is preferred. However, it needs to be evaluated what power offsets are needed and what would be the impact on the overall system performance.

A third alternative is to support only fast intra-Node-B cell selection for HSDPA. However, the potential performance degradation with such a limitation needs to be evaluated.

[1] This information based on long term averaging may be communicated to the UE using L3 messaging from the current cell.

6.4.4 Conclusions

Fast Cell Selection could inherit a significant part of the required physical-layer functionality from SSDT. Thus very little new physical-layer functionality is needed to support Fast Cell Selection. The only identified issue is the possible use of physical-layer signalling to transfer transmission-state between Node B in case of inter-Node-B FCS.

The following should be further evaluated with corresponding conclusions in the RAN1 TR

- *The required UL DPCCH power offset needed to ensure that transmission-state signalling by means of DPCCH can be sufficiently reliably detected at the target Node B in case of inter-Node-B FCS and the corresponding impact on system performance.*
- *The potential performance degradation if fast cell selection is limited to fast intra-Node-B cell selection.*

6.5 Multiple Input Multiple Output Antenna Processing

6.6 Fast scheduling <physical layer interaction>

6.7 Associated signaling needed for operation of High Speed Downlink Packet Access

- It is yet to be determined what part of the associated uplink and downlink signaling is to be physical-layer and what part is to be higher-layer signaling, depending on signaling requirements such as periodicity, transmissions delay, and robustness.

6.7.1 Associated Uplink signaling

It is yet to be determined what part of the associated uplink signaling is to be physical-layer and what part is to be higher-layer signaling, depending on signaling requirements such as periodicity, transmissions delay, and robustness. Associated uplink physical signaling may include, but may not be restricted to

- Signaling for measurement reports related to fast link adaptation (Adaptive modulation and coding, AMC), if such measurements are to be supported
- Signaling related to fast cell selection
- Signaling related to fast Hybrid ARQ.

This associated uplink physical signaling should be carried on the uplink DPCCH. The main impact on current specification is the need for additional uplink DPCCH slot formats and the possible need for lower spreading factor for the uplink DPCCH ($SF_{UL\text{-}DPCCH} < 256$).

6.7.2 Associated Downlink signaling

It is yet to be determined what part of the associated downlink signalling is to be physical-layer and what part is to be higher-layer signalling, depending on signaling requirements such as periodicity, transmissions delay, and robustness. Associated downlink physical signalling may include, but may not be restricted to:

- identifying the UE(s) to which HSDPA data is transmitted in a given HSDPA TTI.
- identifying which HSDPA codes are assigned to a UE in a given HSDPA TTI (if sharing in the code domain, i.e. code multiplexing is to be supported for HSDPA transmission)
- identifying modulation and coding scheme used for HSDPA transmission in a given HSDPA TTI.
- identifying relative CPICH to DSCH power ratio for a HSDPA transmission in a given HSDPA TTI (specifically for QAM modulation).
- identifying or setting current states of fast Hybrid ARQ

Two alternatives have been proposed for the downlink physical signalling associated with HSDPA transmission:

- The entire set of associated downlink signaling is carried on associated downlink dedicated physical channels
- Part of the associated downlink signaling is carried on an associated downlink shared signaling/control channel.

The selection between these alternatives should be based on an evaluation differences in terms of

- Complexity
- Capacity (interference-limited capacity as well as code-limited capacity)

7 Evaluation of Technologies

7.1 Adaptive Modulation and Coding (AMC)

7.1.1 Performance Evaluation <throughput, delay>

For the HSDPA study item, an AMC scheme using 7 MCS levels as outlined in Table-4 of the Annex A were simulated using a symbol level link simulator. The AMC scheme uses QPSK, 8-PSK and 16 and 64 QAM modulation using R=1/2 and R=3/4 Turbo code and can support a maximum peak data rate of 10.8 Mbps. Analytic throughput results with varying number of MCS levels and with Hybrid ARQ disabled/enabled are shown in Table 1 and Table 2 under different channel conditions. An equal average power scheduler was used when computing the analytic throughput. Also, the following variation in MCS levels are considered for the throughput results:

1. Full 7-MCS
2. 5-MCS without QPSK R=1/4, 8PSK R=3/4
3. 4-MCS without QPSK R=1/4, 8PSK R=3/4, 64QAM R=3/4
4. 3-MCS without QPSK R=1/4, QPSK R=3/4, 8PSK R=3/4, 64QAM R=3/4

Table 1. Analytic throughput without HARQ

	Throughput (Mbps/sector/carrier)				
	System Total	0km (Rice)	1km	3km	30km
Case 1: 7-AMCS	**2.516**	2.95	2.67	2.52	2.11
Case 2: 5-AMCS	**2.365**	2.72	2.49	2.38	2.04
Case 3: 4-AMCS	**2.316**	2.67	2.41	2.31	2.04
QPSK R=1/2	***1.426***	*1.55*	*1.41*	*1.40*	*1.39*

Table 2. Analytic throughput with HARQ (Chase Combining)

	Throughput (Mbps/sector/carrier)				
	System Total	0km (Rice, k=12dB)	1km	3km	30km
Case 1: 7-AMCS	**2.776**	3.16	2.88	2.82	2.45
Case 2: 5-AMCS	**2.772**	3.15	2.88	2.83	2.45
Case 3: 4-AMCS	**2.716**	3.08	2.81	2.75	2.40
Case 4: 3-AMCS	**2.5**	-	-	2.5	-
QPSK R=1/2	***1.739***	*1.87*	*1.74*	*1.72*	*1.69*

Figure 3 and Figure 4 show the area probability for both 3 kmph and 120 kmph, respectively. Figure 5 shows the probability of choosing different MCS levels with 100 UEs per sector and using 20% overhead based on system simulation with parameters as per the Annex-A.

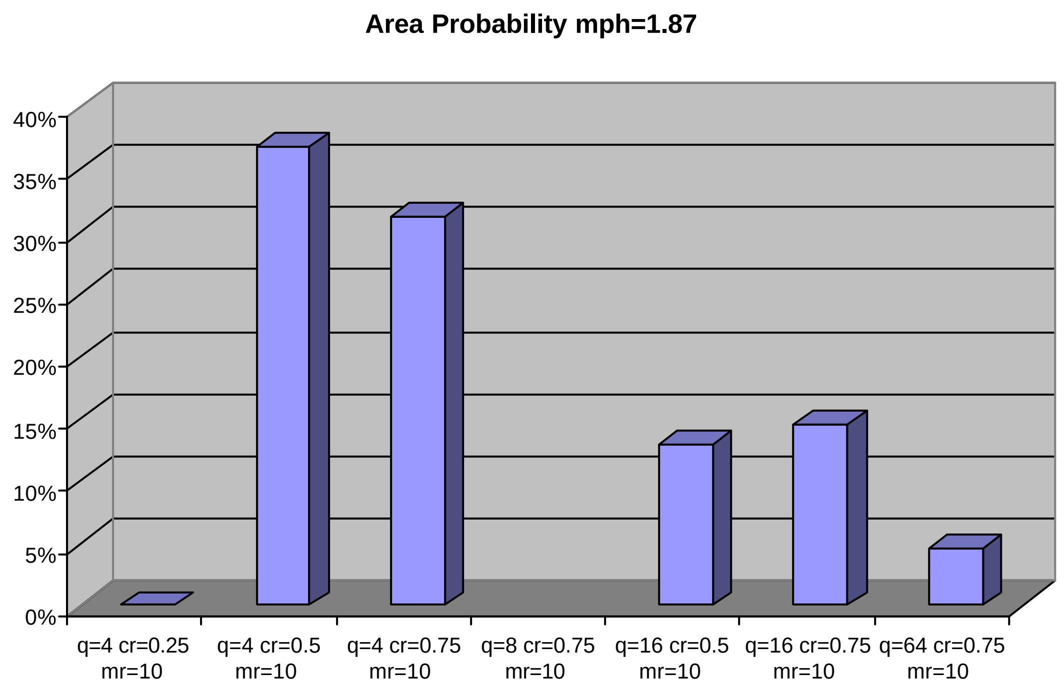


Figure 3. Area probability at 3 Kmph

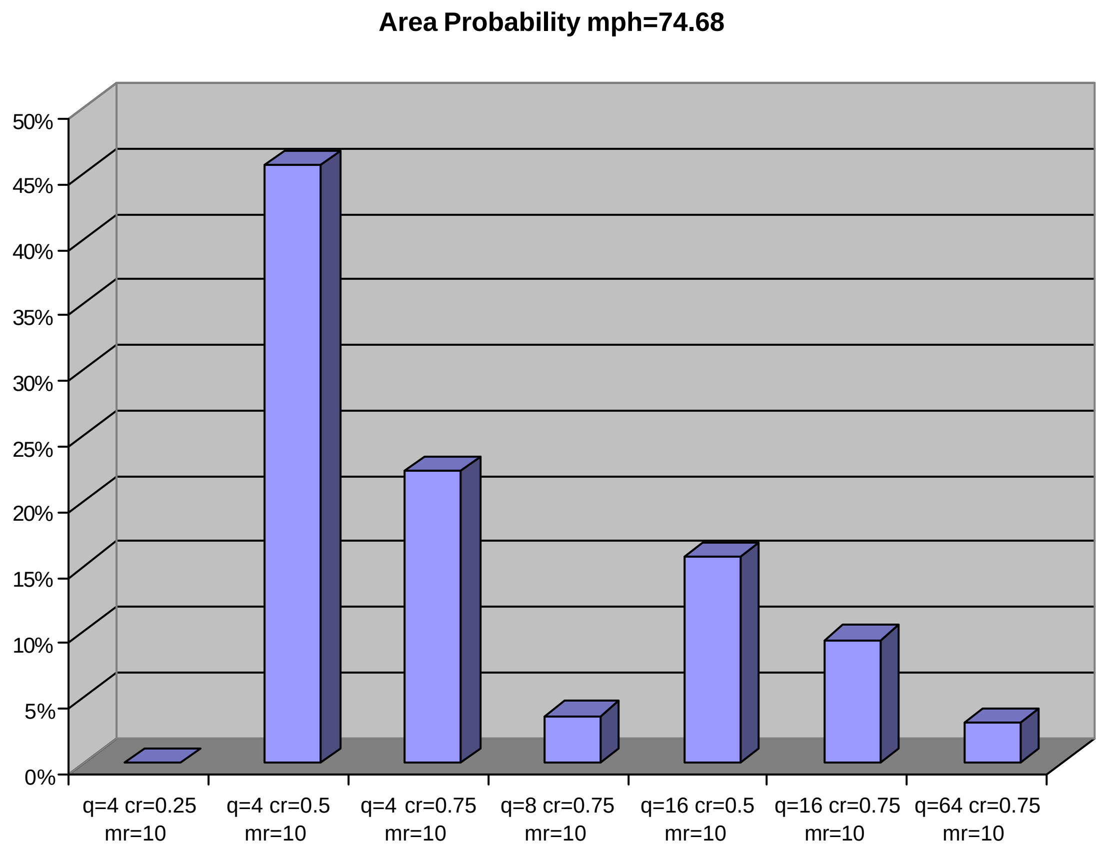


Figure 4. Area probability at 120 Kmph

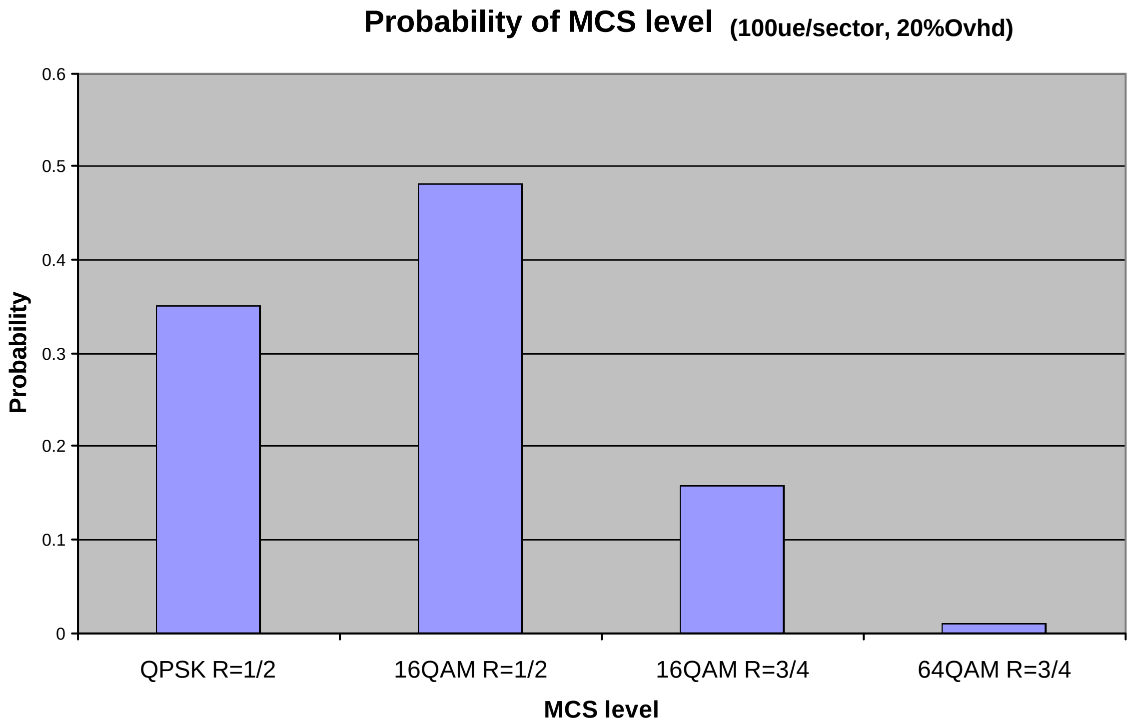


Figure 5. Probability of choosing MCS level (system simulation)

Conclusions:
It may be noted that the MCS level was not varied during re-transmissions in the link and system level simulations for HARQ. Further, it can be concluded from the simulation results that there is a benefit in throughput performance in a system with AMC and HARQ scheme.

References:
[1] Motorola, TSGR1#16, R1-00-1241.
[2] Motorola, TSGR1#17, R1-00-1395.
[3] Sony, TSGR1#17, R1-00-1377.

7.1.2 Complexity Evaluation <UE and RNS impacts>

7.2 Hybrid ARQ (H-ARQ)

7.2.1 Performance Evaluation <throughput, delay>

7.2.2 Complexity Evaluation <UE and RNS impacts>

7.2.2.1 N-channel stop-and-wait H-ARQ

7.2.2.1.1 Introduction

The complexity of H-ARQ mechanisms when employed for link adaptation in HSDPA transmission depends on the H-ARQ scheme selected as well as on where the retransmission functionality is located in the UTRAN. Dual-channel stop-and-wait (SAW) protocol has been proposed as the retransmission functionality for HDSPA. A complexity evaluation on SAW H-ARQ is presented in this section. In this complexity evaluation it is further assumed that H-ARQ retransmission protocol operates in Node B.

7.2.2.1.2 Buffering complexity

The principle of hybrid ARQ is to buffer HSDPA TTIs that were not received correctly and consequently combine the buffered data with retransmissions. The actual method of doing soft combining depends on the H-ARQ combining scheme selected. In Chase combining scheme the receiver always combines the full retransmission of the failed HSDPA TTI, i.e. the amount of data in the receiver buffer remains the same. In the incremental redundancy schemes the receiver buffers coded symbols, which introduce new information to the HSDPA TTI transmitted first, i.e. the amount of data to be buffered increases with consecutive retransmissions. However, probably in practice the buffer in the receiver needs to be dimensioned considering the maximum size of the HSDPA TTI after all the incremental redundancy has been introduced. Regardless of the H-ARQ combining scheme soft combining is done on L1 before the decoding stage of FEC. Prior to decoding these symbols are soft-valued, i.e. each symbol is represented by two or more bits.

Regardless of the location of retransmission functionality in the RNS the number of symbols to be buffered in L1 receiver can be estimated generally as follows:

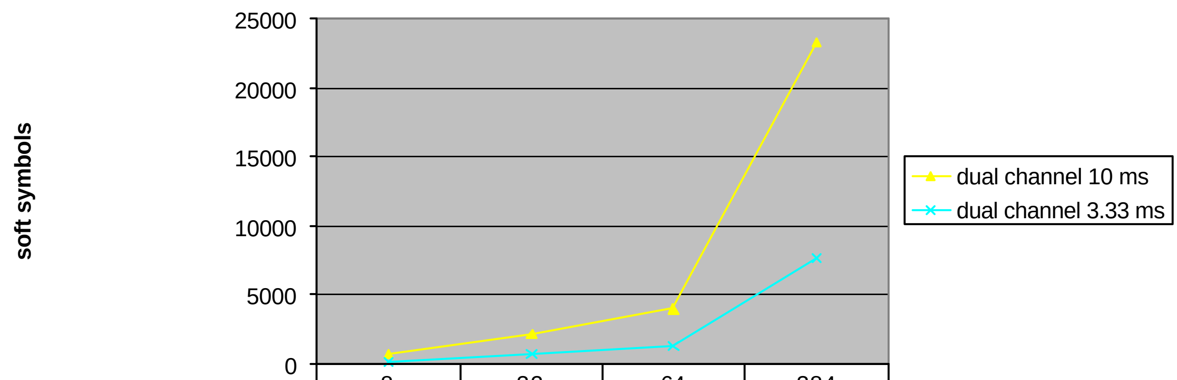
$$buffer = \lceil coded\ bits_{PDU} \cdot failed\ PDUs\ in\ TTI \cdot (latency_{retransmit} + latency_{NACK}) \rceil$$

where it is assumed for the sake of clarity that an integer number of PDUs fit into one HSDPA TTI. The latencies are also considered as multiples of a HSDPA TTI. For dual channel stop-and-wait H-ARQ the buffer size estimation is considerably simplified since no new PDUs are transmitted on a subchannel before the previous packet is acknowledged. The receiver has to buffer one HSDPA TTI from both subchannels. The next transmission is either a new packet or a retransmission of an erroneous packet. In either case, the maximum buffering need is two HSDPA TTIs. The actual size of the buffer needed for each HSDPA TTI depends on the H-ARQ combining scheme as described above. The receiver buffering complexity estimate can be easily extended to *n*-channel stop-and-wait protocol, where at maximum *n* HSDPA TTIs would be buffered at any given time. Thus, for *n*-channel stop-and-wait ARQ the L1 buffering can be expressed as:

$$buffer = \lceil coded\ bits_{TTI} \cdot n \rceil$$

However, it must be noted that the size of HSDPA TTI may change when the number of subchannels changes, i.e. TTI length for n-channel SAW HARQ can be shorter than one for dual channel SAW HARQ. Average receiver buffer sizes for dual channel HARQ for some bit rates are depicted in `Figure 6`.

Naturally, the number of subchannels in stop-and-wait ARQ is reflected in the amount of acknowledgment signaling needed to be sent to the transmitter. The complexity impact on RNS is mainly concentrated on Node B where the H-ARQ retransmission resides according to the current proposal. However, packet buffering is not as much an issue in Node B hardware.



	8	32	64	384
dual channel 10 ms	672	2112	4032	23232
dual channel 3.33 ms	224	704	1344	7744

kbps

Figure 6. Average Receiver L1 buffer size for dual channel SAW HARQ

7.2.2.1.3 Encoding/decoding and rate matching complexity

In order to facilitate incremental redundancy it is likely that the FEC encoder rate has to be lowered, i.e. instead of a 1/3 rate encoder, a 1/5 or even lower rate encoder would be employed. For example, as proposed this far, by puncturing different symbols out of the output code word, different redundancy information is generated for soft combining. A mother code of lower rate does increase the complexity of both encoding and decoding stage.

However, it is not necessary to add new constituent encoders to a turbo coder in order to lower the coding rate. More advanced methods that output more than one symbol per bit per branch could be utilized. Furthermore, investigations are needed to check whether the existing rate matching algorithm of Rel –99 can be used in conjunction with incremental redundancy or whether modification of either the rate matching or the encoder are necessary"

7.2.2.1.4 UE and RNS processing time considerations

References:

[1] Nokia, TSGR1#17, R1-00-1464.

7.3 Fast Cell Selection (FCS)

7.3.1 Performance Evaluation <throughput, delay>

7.3.2 Complexity Evaluation <UE and RNS impacts>

7.4 Multiple Input Multiple Output Antenna Processing

7.4.1 MIMO performance evaluation

Link level simulations were performed and the frame error rate (FER) versus Eb/N0 were measured for a variety of system architectures. We first compare the systems for a fixed data rate and show that, compared to the conventional transmitter, MIMO architectures can achieve the same frame error rate at much lower Eb/N0. Next, we show how for a similar Eb/N0, the MIMO architectures can achieve higher data rates.

The data rate was fixed at 10.8 Mbps, achieved assuming a chipping rate of 3.84 Mchips/sec, a spreading factor of 32 chips per coded symbol, N = 20 spreading codes, and appropriate coding rates and data constellation sizes. A serially concatenated convolutional coding and turbo decoding with 8 decoding iterations [**Editors note: as per the assumptions parallel concatenated Turbo codes should be used]** was used**.** The system architectures for M transmit antennas and P receive antennas are given in Table 3.

Puncturing for the (4,4) system is used to achieve 10.8 Mbps. A flat fading channel with 3km/hr fading, perfect channel estimation, and uncorrelated fading between antenna pairs for the MIMO systems is assumed. Figure 7 below shows the FER versus Eb/N0. Compared to the conventional transmitter, there are gains of about 9dB and 16dB for the (2,2) and (4,4) systems, respectively, at 10% FER. The enormous performance gains are due to a combination of diversity, receiver combining gain, and increased spectral efficiency due to MIMO processing. We emphasize that these gains are achieved using the same code resources (20 codes) as the conventional transmitter.

Table 3. System Architecture for achieving 10.8 Mbps

(M, P)	Tx technique	Code rate	Modu-lation	Rate per substream	Number of substreams	Total data rate
(1,1)	Conven-tional (2x1)	¾	64QAM	540 Kbps	20	10.8Mbps
(2,2)	MIMO	¾	8PSK	270 Kbps	40	10.8Mbps

(4,4)	MIMO	~½	QPSK	135 Kbps	80	10.8Mbps

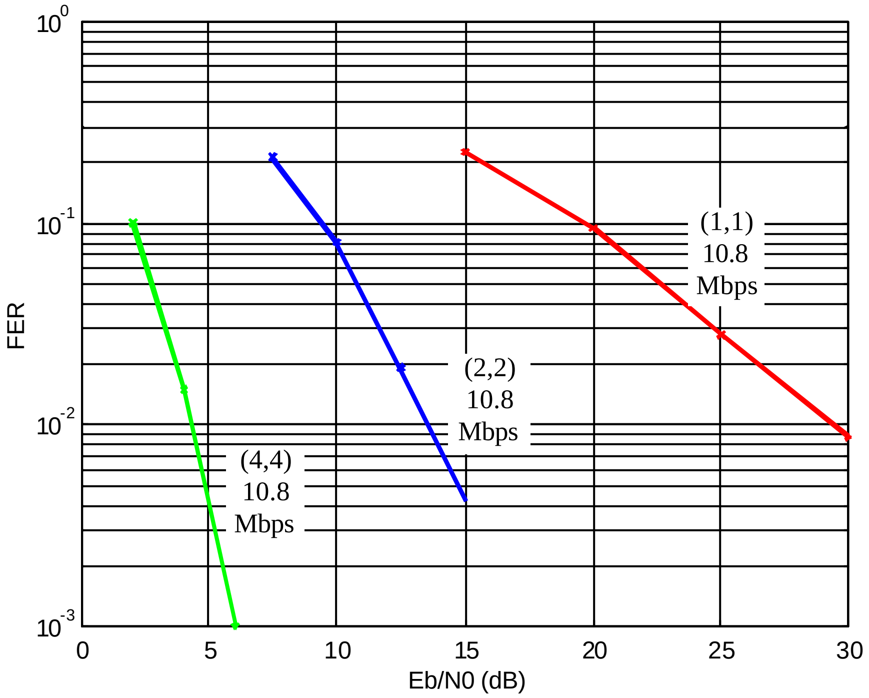


Figure 7. Flat fading channel performance for 10.8 Mbps

Using MIMO techniques, the maximum data rate can increase to 14.4 for the (2,2) system and up to 21.6 Mbps for the (4,4) system. As shown in the Table 4, the constellation sizes are still smaller than those of the conventional transmitter. As seen in Figure 8, the required Eb/N0's for these rates are less than that for the conventional system operating at 10.8Mbps.

Table 4. System Architecture for achieving 21.8 Mbps

(M, P)	Tx technique	Code rate	Modu-lation	Rate per substream	Number of substreams	Total data rate
(1,1)	Conven-tional (2x1)	¾	64QAM	540 Kbps	20	10.8Mbps
(2,2)	MIMO	¾	16QAM	360 Kbps	40	14.4Mbps
(4,4)	MIMO	¾	QPSK	180 Kbps	80	14.4Mbps
(4,4)	MIMO	¾	8PSK	540 Kbps	80	21.6Mbps

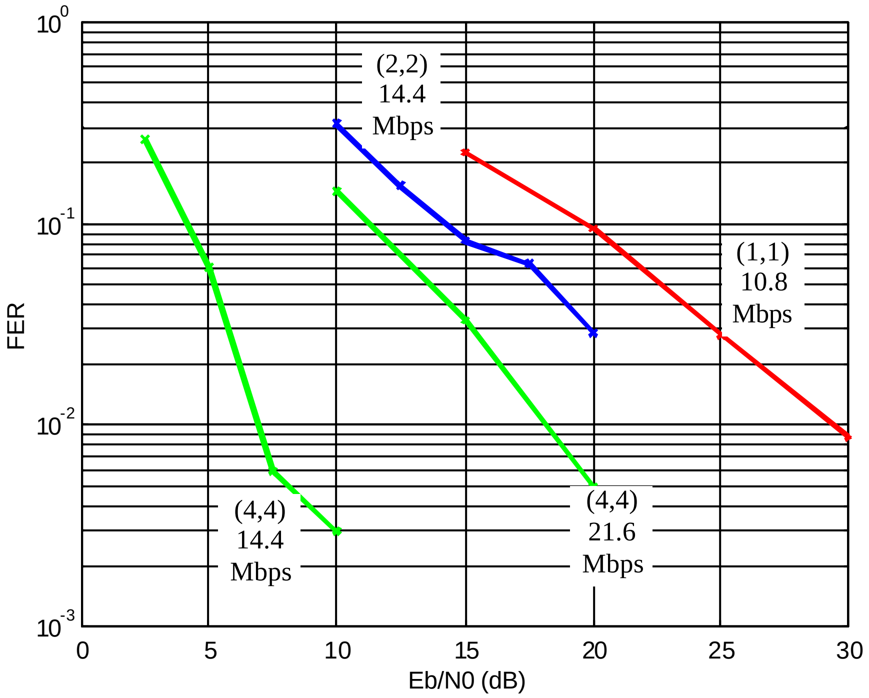


Figure 8. Flat fading channel performance for higher data rates

One way to interpret the Eb/N0 gains for MIMO is that the high data rates can be achieved with less transmit power. Alternatively, if the DSCH is transmitted at a fixed power, then the MIMO gains translate into the higher data rates being used over a larger fraction of the cell area. Under this assumption of a rate-controlled DSCH, a system level study employing a base station scheduler showed that the average sector throughput using a (4,4) MIMO system increases by a factor of 1.8 and 2.8 for proportional fair and maximum C/I scheduling, respectively, compared to a conventional (1,1) system [1]. (As an aside, a surprising result in [1] is that under a proportional fair scheduler, the conventional (1,1) system actually outperforms both the (2,1) and (4,1) diversity systems when there are multiple users vying for the DSCH.). It may be noted that the system level simulation did not use all the assumption as outlined in Annex A.

Additional link level studies investigated the effect of higher doppler frequencies, channel estimation, and correlated channels on the MIMO performance [2]. At 10% FER, compared to a baseline with low doppler, perfect channel estimation, and uncorrelated channels, the worst case loss in required Eb/N0 is about 2dB, occurring in an indoor correlated channel with channel estimation. Hence with regard to these impairments, the MIMO architecture is a robust technique for providing high speed downlink packet service in excess of 10.8 Mbps.

7.4.2 MIMO UE complexity evaluation (Not yet discussed)

The MIMO UE will require multiple uncorrelated antennas and additional baseband processing to perform space-time combining and spatial processing on substreams which share the same code.

High spectral efficiencies of the MIMO system are achieved when there is uncorrelated fading among pairs of transmitter and receiver antennas. Because the terminal receiver is at the same level as local scatterers, only ½ wavelength antenna spacing is required to achieve uncorrelated fading [3]. With 2GHz carrier frequency, an array of 4 antennas with dual-polarization requires only 7.5cm of linear space. High data rate services will most likely target terminals with relatively large screens such as personal digital assistants (PDAs) or laptop computers. Hence these devices will have ample surface area to easily support up to 4 antennas.

A high level investigation on the feasibility of a MIMO baseband processor was summarized in [4]. The main baseband components are a despreader, a space-time combiner, a detector for eliminating spatial interference, and a turbo decoder. It was shown that the turbo decoder requires the majority of the processing power. For a 2 transmit antenna, 2 receive antenna system, the detector portion accounts for about 5% of the total processing and the turbo decoder accounts for about 85% of the processing. For a 4 transmitter, 4 receiver system, the detector and turbo decoder account for about 20% and 70% respectively of the total processing. Compared to a conventional single antenna receiver for HSDPA which requires about 1.6×10^9 operations per second, the 2 antenna receiver requires 3.3×10^9 operations per second and the 4 antenna receiver requires 7.9×10^9 operations per second. These computational requirements were estimated assuming brute-force processing techniques but are already within the range of existing hardware technologies. More detailed studies will mostly likely reduce the processing requirements significantly.

7.4.3 Node-B Complexity Evaluation

References

[1] Lucent. Throughput simulations for MIMO and transmit diversity enhancements to HSDPA. TSG_R WG1 document TSGR1#17(00)1388, 21-24th, November 2000, Stockholm, Sweden.

[2] Lucent. Further link level results for HSDPA using multiple antennas. TSG_R WG1 document TSGR1#17(00)1386, 21-24th, November 2000, Stockholm, Sweden.

[3] D. Chizhik, F. Rashid-Farrokhi, J. Ling, A. Lozano, “Effect of antenna separation on the capacity of BLAST in correlated channels,” to appear in *IEEE Communications Letters*.

[4] Lucent. Practical aspects of multiple antenna architectures for HSDPA. TSG_R WG1 document TSGR1#16(00)1220, 10-13th, October 2000, Pusan, Korea

8 Backwards compatibility aspects

9 Conclusions and recommendations

10 References

[1] Motorola. Link Evaluation Methods for High Speed Downlink Packet Access (HSDPA). TSG-R1 document, TSGR#14(00)0910, 4-7th, July, 2000, Oulu, Finland, 6 pp.

[2]Motorola. Evaluation Methods for High Speed Downlink Packet Access (HSDPA). TSG-R1 document, TSGR#14(00)0909, 4-7th, July, 2000, Oulu, Finland, 15 pp.

[3]Nokia. High Speed Downlink Packet Access simulation assumptions. TSG-R1 document, TSGR#14(00)0881, 4-7th, July, 2000, Oulu, Finland, 9 pp.

[4]ETSI SMG2. Universal Mobile Telecommunications System (UMTS); Selection procedures for the choice of radio transmission technologies of the UMTS. ETSI SMG2 Technical Report, TR 101 112 v3.2.0 (UMTS 30.03), 83 pp.

[5] Gary R. Wright, Richard Stevens. TCP/IP Illustrated, Volume 2: The Implementation;, Addison-Wesley Professional Computing.

11 History

Document history		
Date	**Version**	**Comment**
10/10/00	0.0.0	First draft
10/12/00	0.1.0	Document flow modified and text added with regard to associated signaling and multiple input and multiple output antenna processing.
11/24/00	0.2.0	Text and simulation results added
1/16/01	0.3.0	Revised from Section 7 onwards as per comments received in the plenary
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Tel. : 01-847-632-4121 Fax: 01-847-435-0789 Email :		
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12 Annex A Simulation Assumptions and Results

12.1 Link Simulation Assumptions

12.1.1 Link Assumptions

The objective of this section is to propose a set of definitions, assumptions, and a general framework for performing initial link level simulations for High Speed Downlink Packet Access (HSDPA). The objective of these link level simulations is to provide the needed input data to initial system level simulations and to evaluate the link performance of different Adaptive Modulation and Coding schemes and fast Hybrid ARQ methods.

12.1.2 Simulation Description Overview

A symbol level downlink simulator may be used to simulate the performance of higher order modulation schemes and Hybrid ARQ. The general forward link simulation model is shown in Figure 1. The terminology used throughout the document is as follows: I_{or} is the total transmitted power density by a BTS, $\hat{I}_{or}$ is the post-channel transmitted power density, I_{oc} ? N_o is the other cell interference plus noise power density and I_o is the total received power density at the MS antenna. Note, that the ratio

$\hat{I}_{or}$ /?I_{oc} ? N_o? is fixed in this simulation model. Since the base station has a fixed amount of power (set by the BTS power amplifier size), it is the average transmitted (often called allocated) power by the BTS to the MS that determines the user capacity of the forward link. This fraction of allocated power is called average traffic channel E_c/I_{or} and is inversely proportional to the forward link capacity.

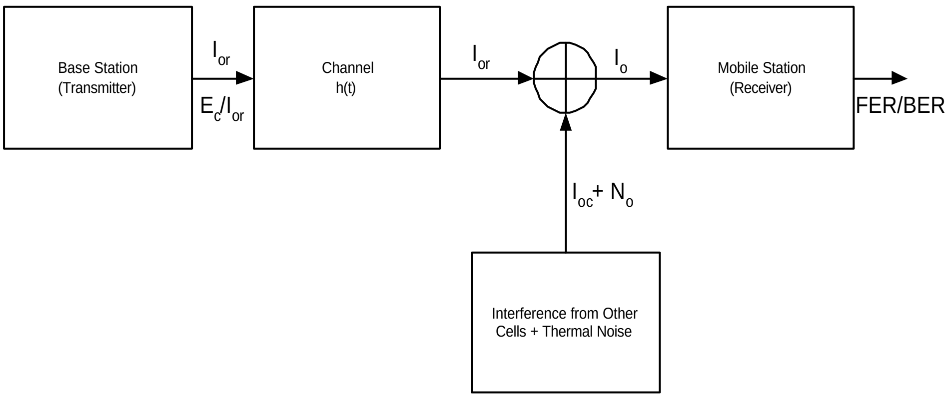

Figure 9. Simulation Block Diagram.

12.1.3 Standard Constellations for M-ary Modulation

In case of 8-PSK modulation, every three binary symbols from the channel interleaver output shall be mapped to a 8-PSK modulation symbol according to Figure 2.

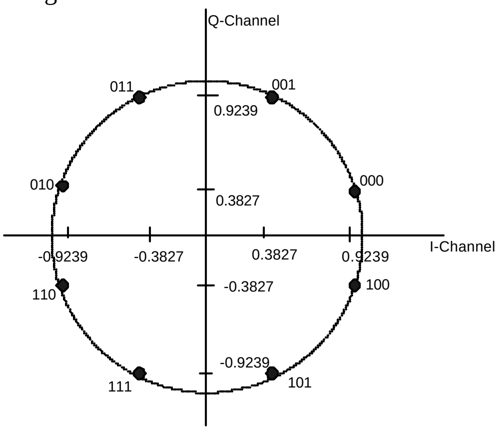

Figure 10. Signal Constellation for 8-PSK Modulation.

In case of 16-QAM modulation, every four binary symbols of the block interleaver output shall be mapped to a 16-QAM modulation symbol according to Figure 3.

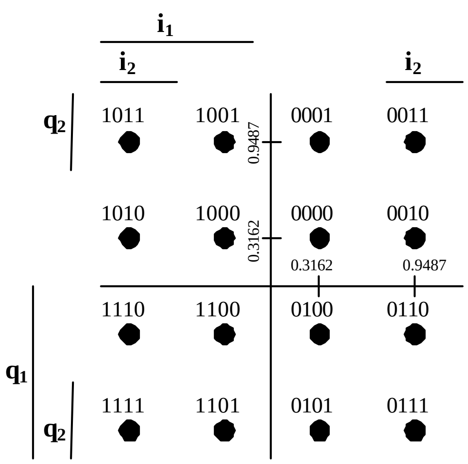

Figure 11. Signal Constellation for 16-QAM Modulation.

In case of 64-QAM modulation, every six binary symbols of the block interleaver output shall be mapped to a 64-QAM modulation symbol according to Figure 4.

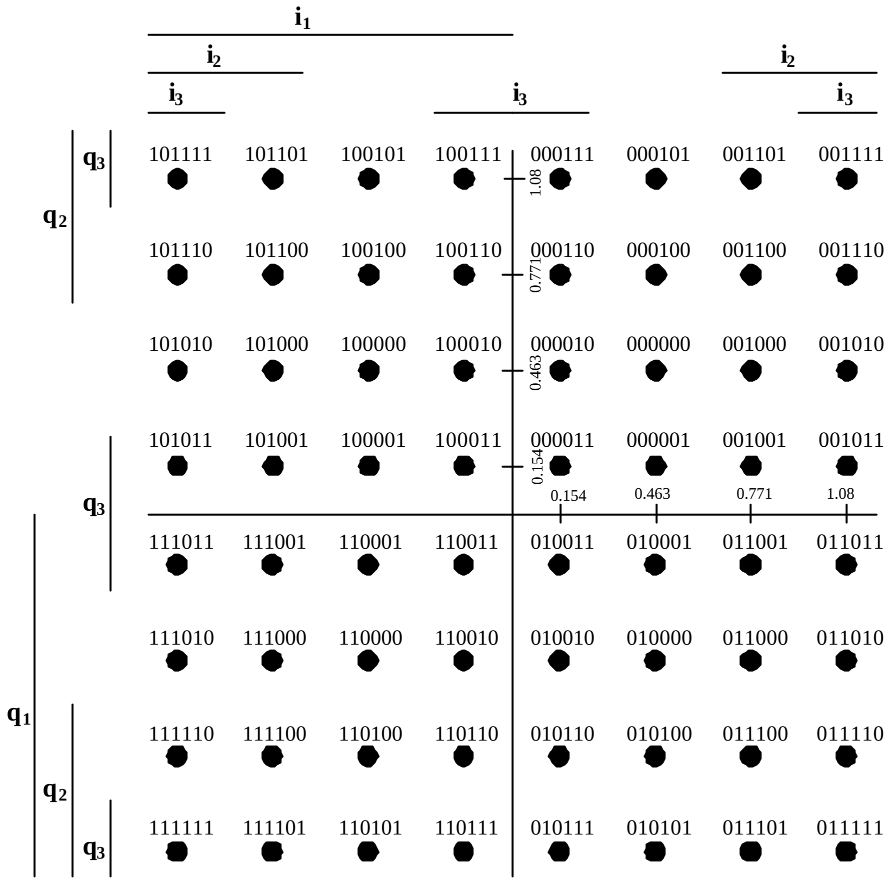

Figure 12. Signal Constellation for 64-QAM Modulation.

12.1.4 Turbo decoding

The *M*-ary QAM demodulator generates soft decisions as inputs to the Turbo decoder. As a baseline method, the soft inputs to the decoder may be generated by an approximation to the log-likelihood ratio function. First define,

$$\Lambda^{(i)}(z) = K_f \left\{ \underset{j \in S_i}{Min}\{d_j^2\} - \underset{j \in \overline{S}_i}{Min}\{d_j^2\} \right\}, \quad i = 0,1,2,\ldots,\log_2 M - 1 \qquad (1)$$

where *M* is the modulation alphabet size, i.e. 8, 16, 32 or 64 and

$$z = A_d A_p \rho \hat{\rho} e^{j(\theta - \hat{\theta})} x + n, \qquad (2)$$

x is the transmitted QAM symbol, A_d is the traffic channel gain, A_p is the pilot channel gain, $\rho e^{j\theta}$ is the complex fading channel gain, and $A_p \hat{\rho} e^{j\hat{\theta}}$ is the fading channel estimate obtained from the pilot channel,

$$S_i = \{ j : i^{th} \text{ component of } y_j \text{ is "0"} \}, \qquad (3)$$

$$\overline{S_i} = \{ j : i^{th} \text{ component of } y_j \text{ is "1"} \} \qquad (4)$$

and K_f is a scale factor proportional to the received signal-to-noise ratio. The parameter d_j is the Euclidean distance of the received symbol z from the points on the QAM constellation in S or its complement. The Pilot/Data gain is assumed known at the receiver. In this case the distance metric is computed as follows

$$d_j^2 = \left| A_p z - Q_j \hat{\rho}^2 \right|^2 \quad Q_j \in S_i \ \textbf{or} \ \overline{S}_i \qquad (5)$$

where $? \; ? \; A_d$ and $? \; ? \; A_p ?$ is an estimate formed from the pilot channel after processing through the channel estimation filter.

12.1.5 Other Decoding

12.1.6 Performance Metrics:

The following link performance criteria are used:

1. FER vs. E_c / I_{or} (for a fixed $\hat{I}_{or} / ?I_{oc} \; ? \; N_o ?$) or

FER vs. $\hat{I}_{or} / ?I_{oc} \; ? \; N_o ?$ (for a fixed E_c / I_{or})

2. Throughput vs. E_c/I_{oc}

where throughput measured in term of bits per second : $T \; ? \; R ? \frac{1 ? FER_r}{\overline{N}} ?$ in bits per second

where T is the throughput, R is the transmitted information bit rate and FER_r is the residual Frame Error Rate beyond the maximum number of transmissions and $\overline{N}$ is the average number of transmission attempts.

12.1.7 Simulation Parameters:

Table 1. provides a list of link-level simulation parameters.

Table 5. Simulation Parameters

Parameter	Value	Comments
Carrier Frequency	2GHz	
Propagation conditions	AWGN, Flat, Pedestrian A (3 Kmph)	Additional channel cases?
Vehicle Speed for Flat Fading	3 kmph/30 kmph/120 kmph	
CPICH relative power	10% (-10dB)	
Closed loop Power Control	OFF	Power control may be used for signalling channels associated with HSDPA transmission
HSDPA frame Length[1]	10ms, 3.33 ms, 0.67 ms	
Ior/Ioc	Variable	
Channel Estimation	Ideal/Non-Ideal(using CPICH)	
Fast fading model	Jakes spectrum	Generated e.g. by Jakes or filtering approach
Channel coding	Turbo code (PCCC), rate 1/4, 1/2, 3/4, etc.	
Tail bits	6	
Max no. of iterations for Turbo Coder	8	
Metric for Turbo Coder	Max[2]	
Input to Turbo Decoder	Soft	
Turbo Interleaver	Random	
Number of Rake fingers	Equal to number of taps in the channel model	
Hybrid ARQ	Chase combining	For initial evaluation of fast HARQ. Other schemes may also be studied.
Max number of frame transmissions for H-ARQ		Specify the value used
Information Bit Rates (Kbps)	As defined	
Number of Multicodes Simulated	As defined	
TFCI model	Random symbols, ignored in the receiver but it is assumed that the receiver gets error free reception of TFCI information	
STTD	On/Off	
Other L1 Parameters	As Specified in Release-99 Specification	

Table Table 6 thru Table 8 shows examples of numerology for HSDPA frames of length 0.67 ms (1 slot), 3.33 ms (5 slots), and 10 ms (15 slots) respectively for different MCS and different number of HSDPA codes[3].

Table 6. Information bit rate for frame duration of 0.67 msec

Chip Rate = 3.84 Mcps				SF = 32			Frame Size = 0.67 ms	
MCS	20 codes			1 code			Code rate	Modulation
	Info Rate	Info bits/frame		Info Rate	Info bits/frame			
	(Mbps)	(bits)	(octets)	(Mbps)	(bits)	(octets)		
7	10.8000	7200	900	0.54	360	45	3/4	64
6	7.2000	4800	600	0.36	240	30	3/4	16

1 According to system simulation assumption document [4], 3.33 msec frame will be prioritized for simulation purpose.

2 Optimum performance can be achieved with max* metric. However, this metric is sensitive to SNR scaling.

3 The transport block size is TBD.

5	4.8000	3200	400	0.24	160	20	1/2	16
4	5.4000	3600	450	0.27	180	22.5	3/4	8
3	3.6000	2400	300	0.18	120	15	3/4	4
2	2.4000	1600	200	0.12	80	10	1/2	4
1	1.2000	800	100	0.06	40	5	1/4	4

Table 7 . Information bit rate for frame duration of 3.33 msec

Chip Rate = 3.84 Mcps				SF = 32			Frame Size = 3.33 ms	
MCS	20 codes			1 code			Code rate	Modulation
	Info Rate (Mbps)	Info bits/frame (bits)	(octets)	Info Rate (Mbps)	Info bits/frame (bits)	(octets)		
7	10.8000	36000	*4500*	0.54	1800	*225*	3/4	64
6	7.2000	24000	*3000*	0.36	1200	*150*	3/4	16
5	4.8000	16000	2000	0.24	800	100	1/2	16
4	5.4000	18000	2250	0.27	900	112.5	3/4	8
3	3.6000	12000	1500	0.18	600	75	3/4	4
2	2.4000	8000	*1000*	0.12	400	*50*	1/2	4
1	1.2000	4000	500	0.06	200	25	1/4	4

Table 8. Information bit rate for frame duration of 10 msec

Chip Rate = 3.84 Mcps				SF = 32			Frame Size = 10.00 ms	
MCS	20 codes			1 code			Code rate	Modulation
	Info Rate (Mbps)	Info bits/frame (bits)	(octets)	Info Rate (Mbps)	Info bits/frame (bits)	(octets)		
7	10.8000	1E+05	13500	0.54	5400	675	3/4	64
6	7.2000	72000	9000	0.36	3600	450	3/4	16
5	4.8000	48000	6000	0.24	2400	300	1/2	16
4	5.4000	54000	6750	0.27	2700	337.5	3/4	8
3	3.6000	36000	4500	0.18	1800	225	3/4	4
2	2.4000	24000	3000	0.12	1200	150	1/2	4
1	1.2000	12000	1500	0.06	600	75	1/4	4

12.1.8 Simulation Cases

12.2 Link Simulation Results

For the HSDPA study item, an AMC scheme using 7 MCS levels as outlined in Table-4 of the Annex were simulated using a symbol level link simulator. The AMC scheme uses QPSK, 8-PSK and 16 and 64 QAM modulation using R=1/2 and R=3/4 Turbo code and can support a maximum peak data rate of 10.8 Mbps.

Figure 14 to Figure 16 shows some sample FER vs. Ior/Ioc curves for various MCS levels for a fixed power allocation of –1dB. It may be observed that as the order of MCS is increased the Ior/Ioc requirements gets higher to achieve the same FER.

Figure 17 to Figure 19 shows the throughput performance of the AMC at 3 kmph and with Hybrid ARQ (Chase combining) enabled/disabled. Figure 20 to Figure 22 shows the performance of the AMC at 120 kmph and with Hybrid ARQ (Chase combining) enabled/disabled. The following abbreviations are used in the Figures:

CE – Channel Estimation – 0(ideal), 1 (non-ideal)

STTD – Transmit Diversity – 0(off), 1(on)

NC – Number of codes

ARQ – Hybrid ARQ using Chase combining – 0(off), 1(on)

NP – Number of paths

FC – Carrier frequency (2 GHz)
IBM – Ray Imbalance
ECIOR = Power Allocation (set to 80%)
SPEED – Vehicle speed
Q – QAM Modulation Level (4/8/16/64)
CR – Turbo Code Rate
MR – Maximum number of repeats

The following observations are made from the figures:

a. MCS level 1 (R=1/4 using QPSK) and MCS level 3 (R=3/4 using 8 PSK) does not have any throughput advantages for geometries (Ior/Ioc) ranging from 0 dB to 20dB. In view of the above, the number of MCS levels can easily be reduced from 7 to 5.
b. The power requirement doubles as the number of codes are doubled.
c. AMC with HARQ provides higher throughput than AMC without Hybrid ARQ especially at slow speeds. A HARQ system with fast feedback, can ensure that extra redundancy is sent only when needed while still meeting delay constraints.

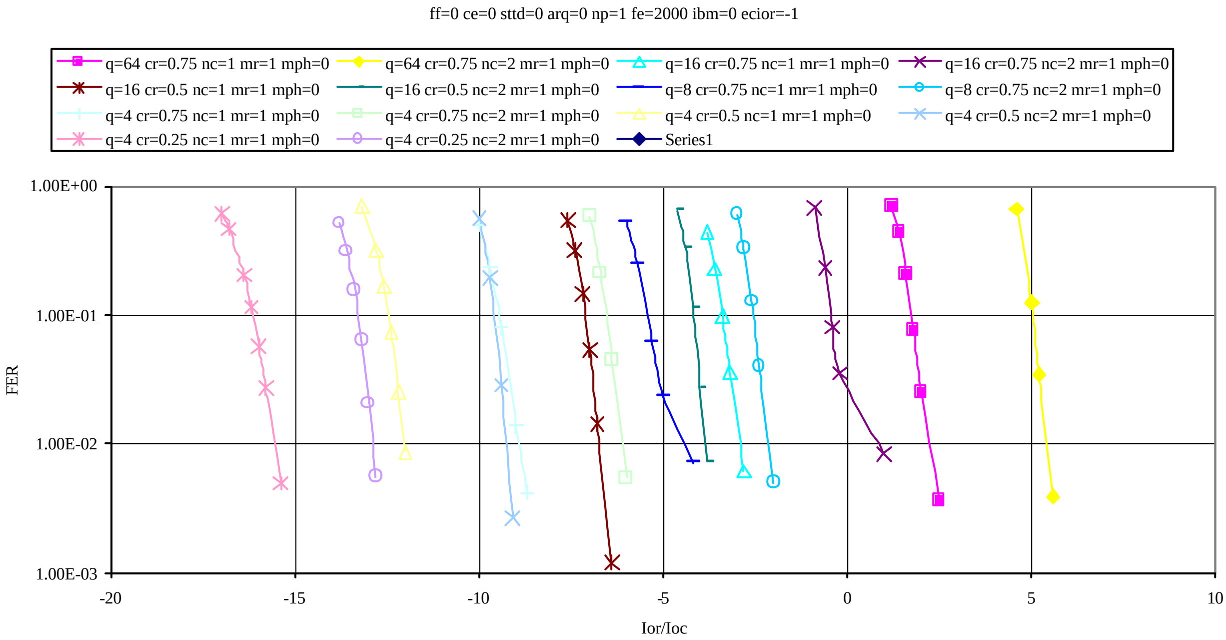

Figure 13. FER vs. Ior/Ioc for various MCS schemes Channel -Static

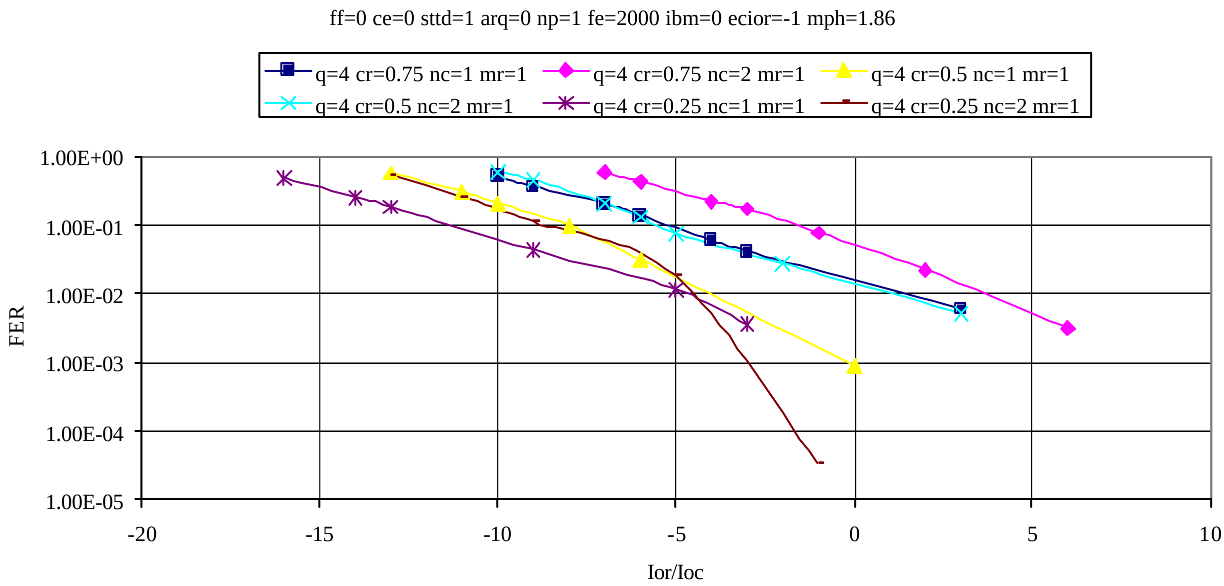

Figure 14. FER vs. Ior/Ioc for QPSK – Flat Fading @3 kmph

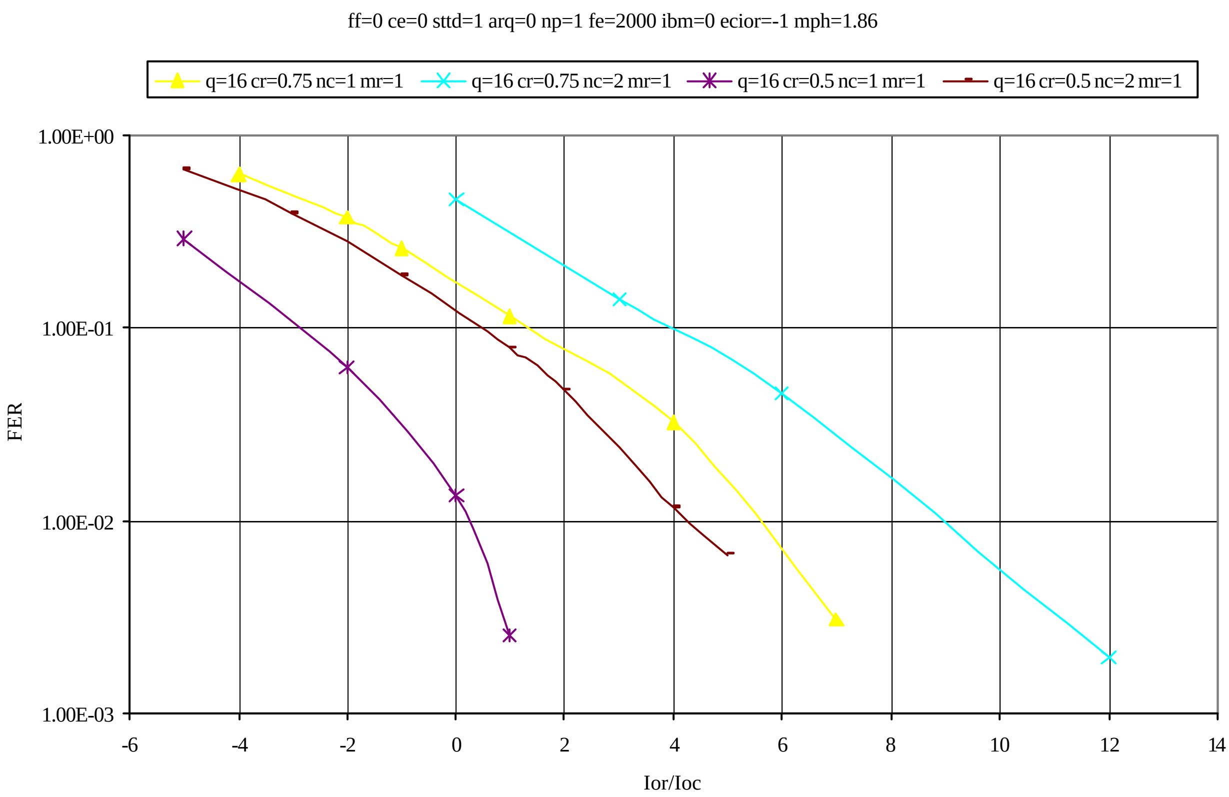

Figure 15. FER vs. Ior/Ioc for 16 QAM– Flat Fading @ 3Kmph

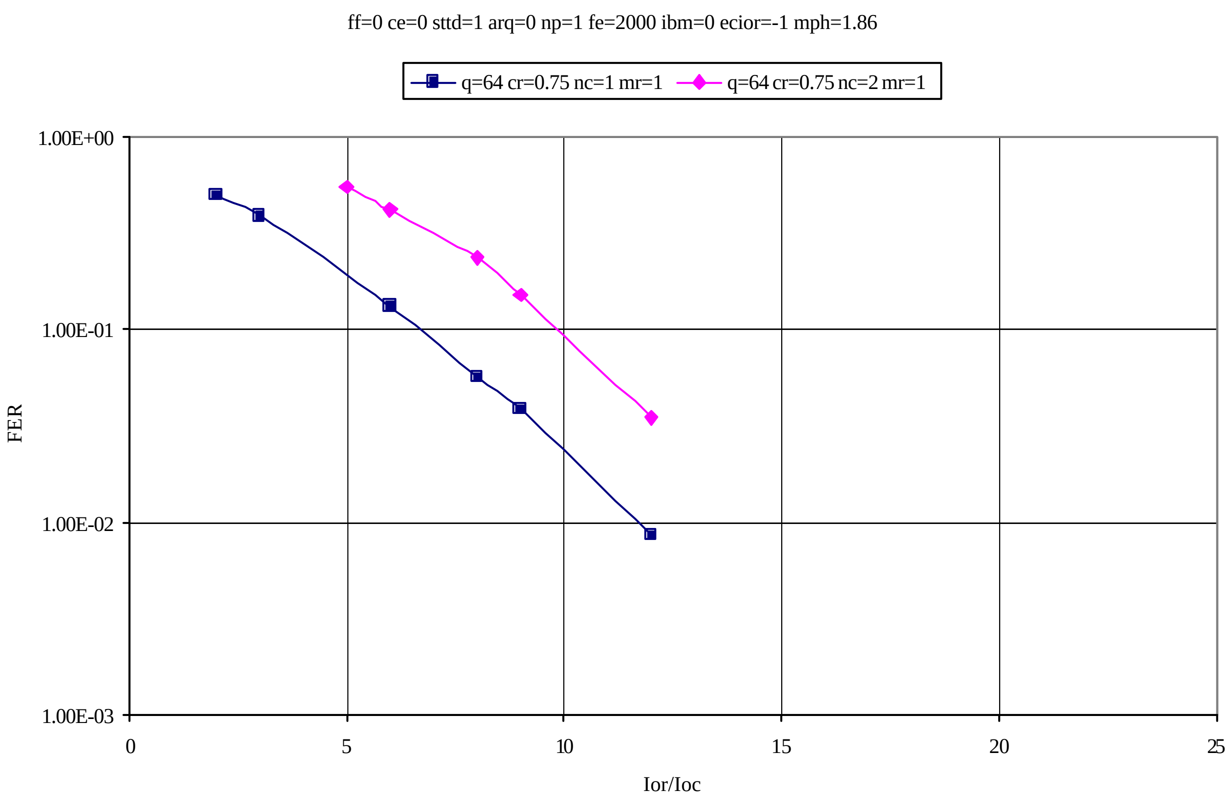

Figure 16. FER vs. Ior/Ioc for 64-QAM – Flat Fading @ 3Kmph

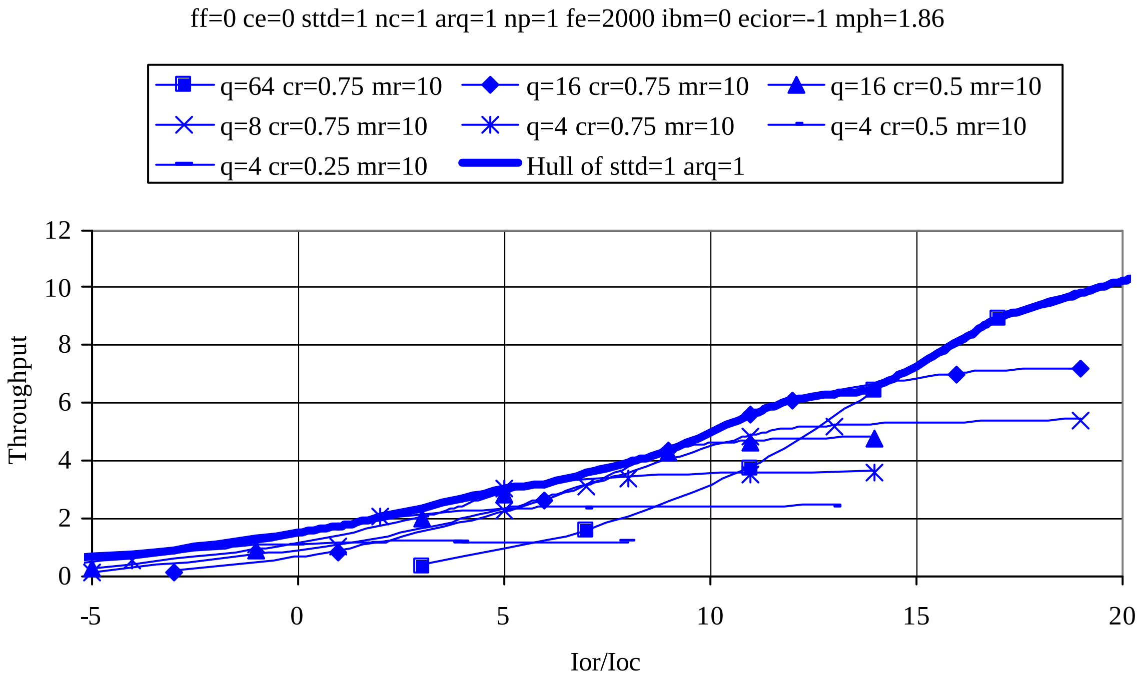

Figure 17. Throughput at 3kmph with HARQ

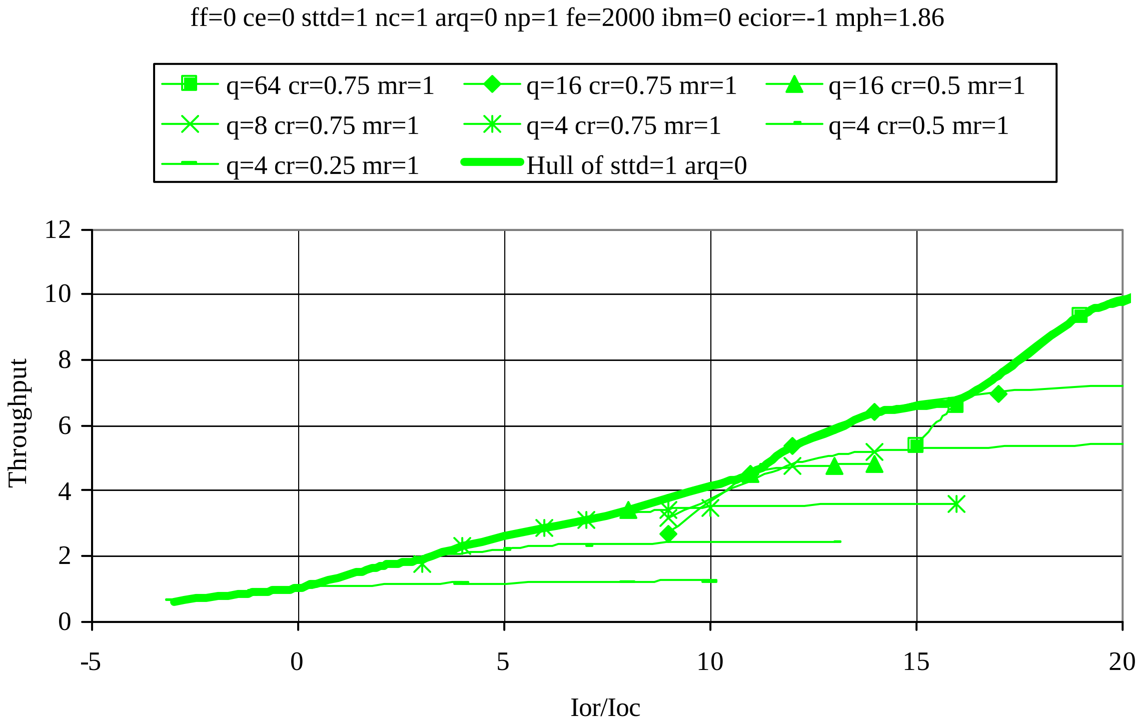

Figure 18. Throughput at 3 kmph without HARQ

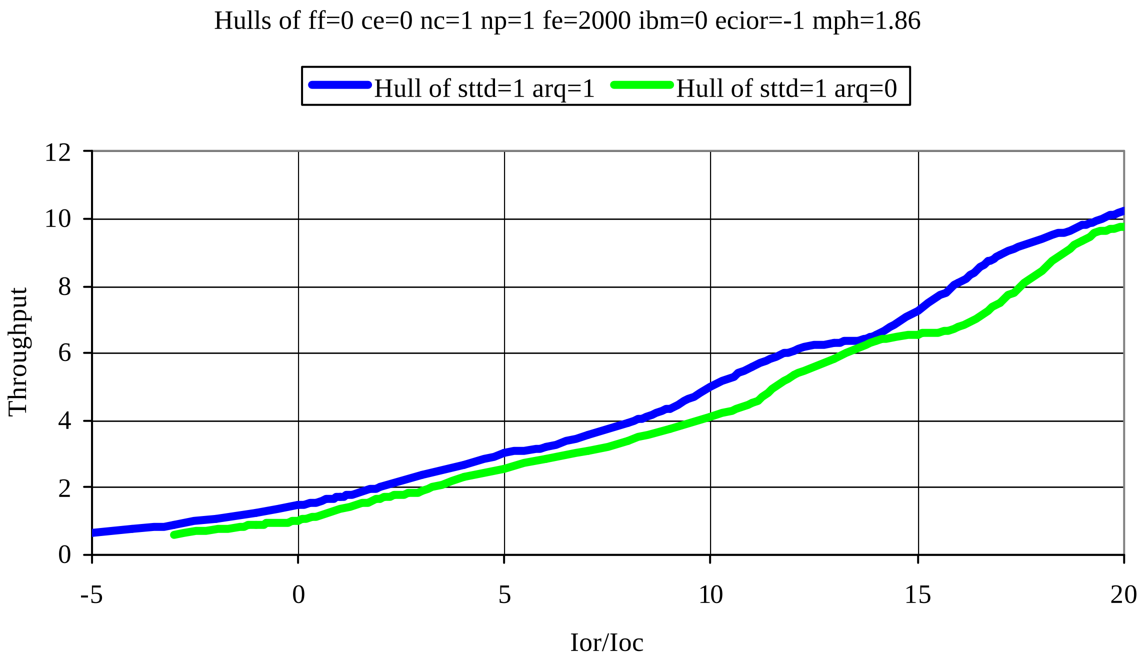

Figure 19. Hull curves at 3 kmph with and without HARQ

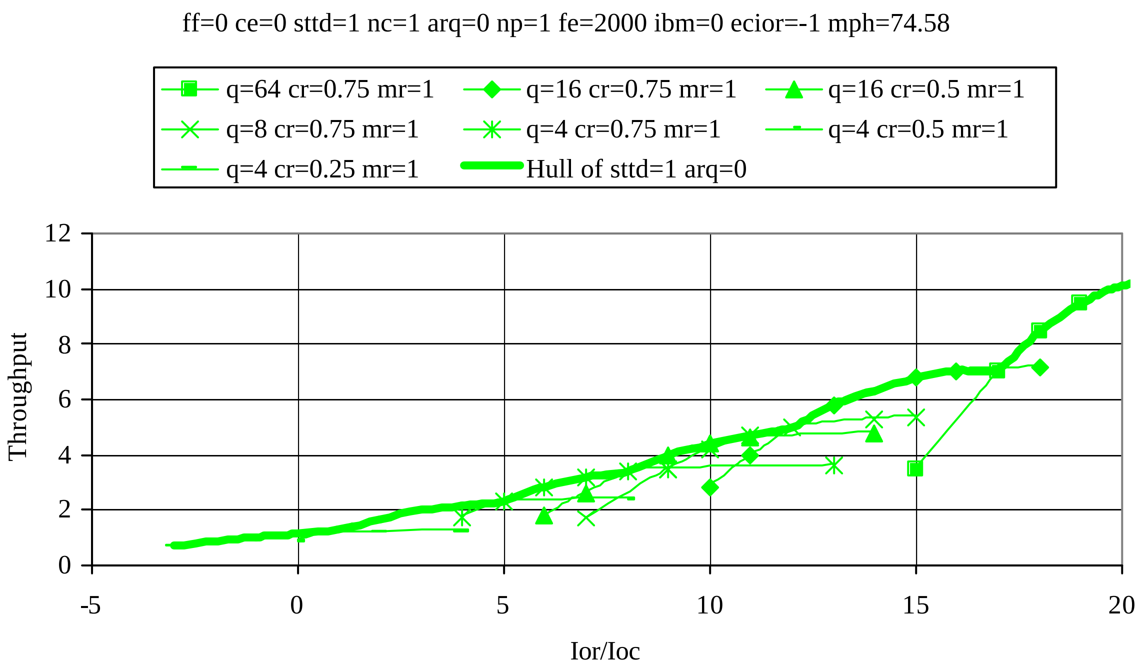

Figure 20. Throughput at 120 kmph without HARQ

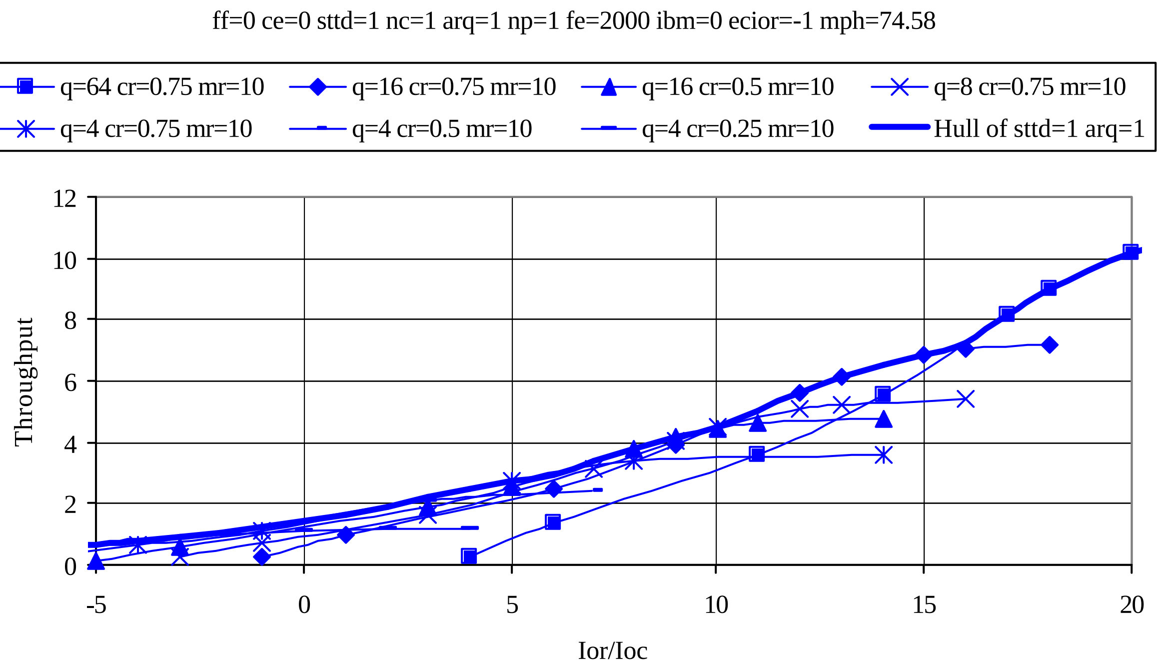

Figure 21. Throughput at 120 kmph without HARQ

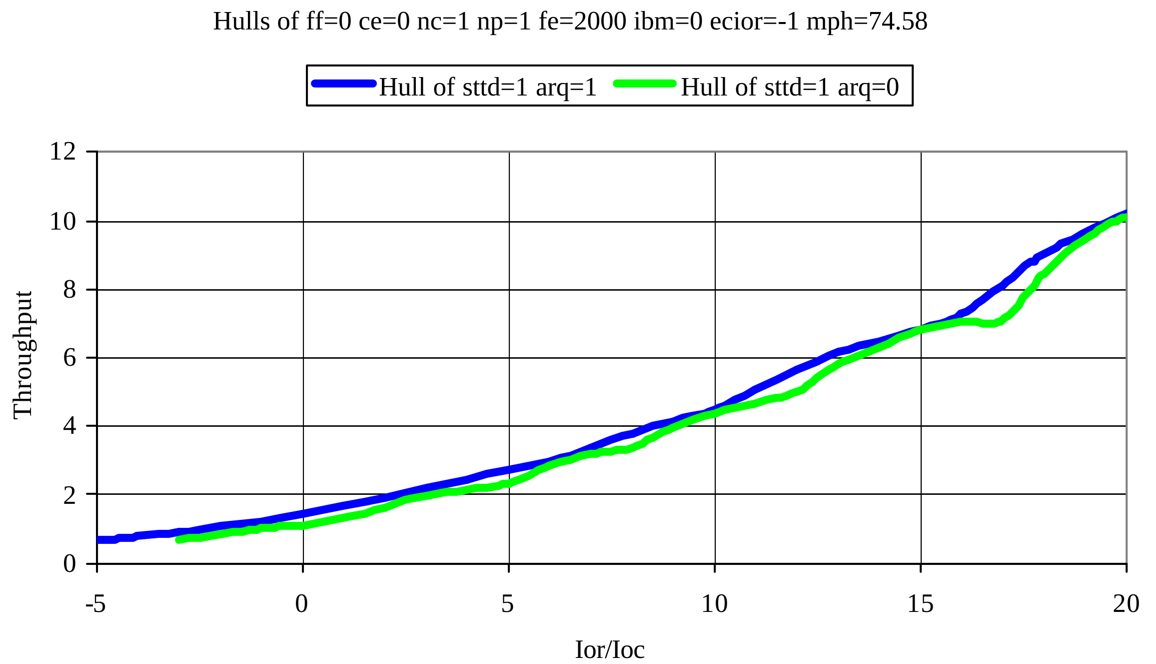

Figure 22. Hull curves at 120 kmph with and without HARQ

12.2.1 Effect of multipath

12.2.2 Effect of non-ideal channel estimation:

Figure 23 and Figure 24 show the FER versus Ior/Ioc for QPSK modulation in a 1-path Rayleigh with vehicle speed of 3kmph and 120kmph respectively. Figure 25 and Figure 26 show the FER versus Ior/Ioc for 16QAM modulation in a 1-path Rayleigh with vehicle speed of 3kmph and 120kmph respectively. Notice that with ideal channel estimation (ICE), the FER performance at higher vehicle speed is better than that at lower vehicle speed for both QPSK and 16QAM due to more un-correlated errors at higher speed. However, for the higher speed case, the channel estimation errors are generally large due to the time-variant behaviour of the channel. Such channel estimation errors may dramatically degrade the performance as shown in the figures. For higher modulation, such as 16QAM and 64QAM, the channel estimation accuracy is more important because of the close signal constellation and both phase and magnitude related soft input calculation.

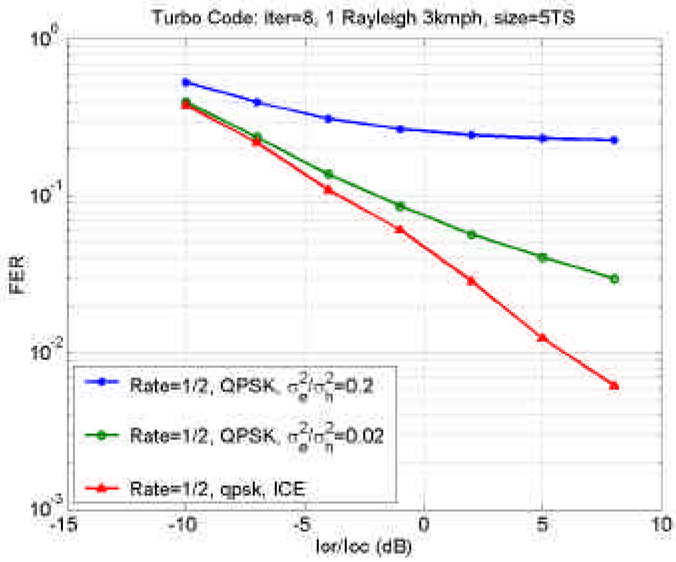

Figure 23. FER versus Ior/Ioc , R=1/2 QPSK, 3 kmph

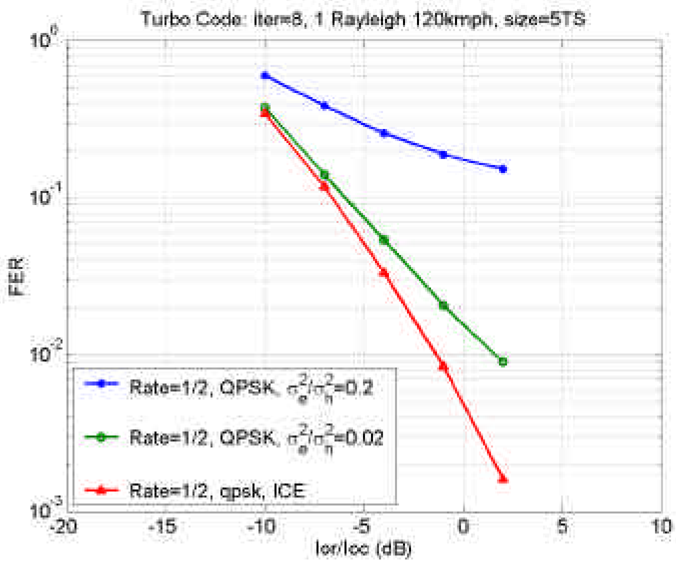

Figure 24 FER versus Ior/Ioc , R=1/2 QPSK, 120 kmph

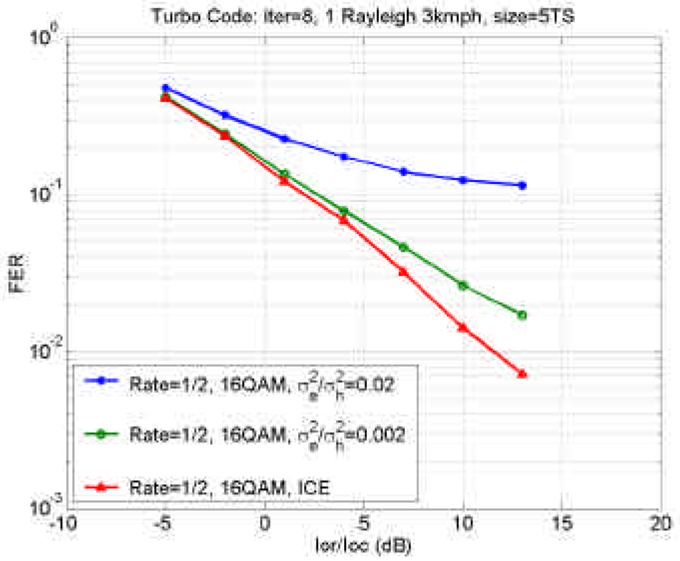

Figure 25 FER versus Ior/Ioc , R=1/2 16 QAM, 120 kmph.

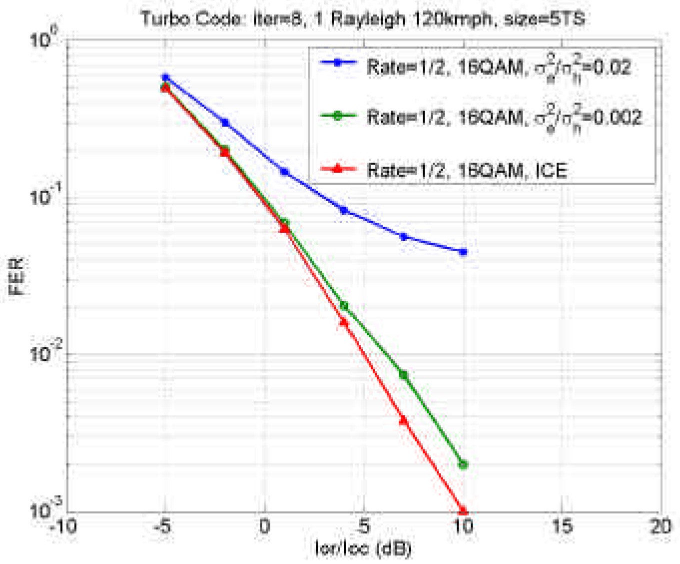

Figure 26 FER versus Ior/Ioc , R=1/2 QAM, 120 kmph.

Conclusion

Preliminary simulation results regarding the effect of channel estimation error on the link level performance of HSDPA is shown. The link level performance of higher order modulation is very sensitive to the channel estimation (used for channel compensation and other purposes) and thus accurate channel estimation is essential, especially at high vehicle speeds. More investigation on this topic and novel techniques to handle this practical issue are needed.

References:

[1] Motorola, TSGR1#16, R1-00-1241.
[2] Motorola, TSGR1#17, R1-00-1395.
[3] Sony, TSGR1#17, R1-00-1377.
[4] WISCOM, TSGR1#17, R1-00-1326.
[5] WISCOM, TSGR1#17, R1-00-1327

12.3 System Simulation Assumptions

The scope of this section is to propose a set of definitions and assumptions on which HSDPA simulations can be based. The initial objective of such system simulations should be to illustrate/verify the potential performance gains due to the currently proposed HSDPA features, such as adaptive modulation and coding scheme (AMCS), fast Hybrid ARQ, and fast cell selection (FCS).

12.3.1 Common System Level 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. In Annex 1, two different methods are outlined. They should be seen as examples and therefore other methods can be used.

12.3.2 Basic system level parameters

The basic system level simulation parameters are listed in Table 5 below.

Table 9. Basic system level simulation assumptions.

Parameter	Explanation/Assumption	Comments
Cellular layout	Hexagonal grid, 3-sector sites	Provide your cell layout picture
Site to Site distance	2800 m	
Antenna pattern	As proposed in [2]	Only horizontal pattern specified
Propagation model	$L = 128.1 + 37.6\ Log_{10}(R)$	R in kilometers
CPICH power	-10 dB	
Other common channels	- 10 dB	
Power allocated to HSDPA transmission, including associated signaling	Max. 80 % of total cell power	
Slow fading	As modeled in UMTS 30.03, B 1.4.1.4	
Std. deviation of slow fading	8 dB	
Correlation between sectors	1.0	
Correlation between sites	0.5	
Correlation distance of slow fading	50 m	
Carrier frequency	2000 MHz	
BS antenna gain	14 dB	
UE antenna gain	0 dBi	
UE noise figure	9 dB	
Max. # of retransmissions	Specify the value used	Retransmissions by fast HARQ
Fast HARQ scheme	Chase combining	For initial evaluation of fast HARQ
BS total Tx power	Up to 44 dBm	
Active set size	3	Maximum size
Specify Fast Fading model	Jakes spectrum	Generated e.g. by Jakes or Filter approach

12.3.3 Data traffic model

The described data-traffic model simulates bursty web traffic. The parameters of the model are based on [4] but have been tailored to reduce simulation run time by decreasing the number of UEs required to achieve peak system loading. The main modification is to reduce the reading time between packet calls. In addition, TCP/IP rate adaptation mechanisms have been included to pace the packet arrival process of packets within a packet call.

The model assumes that all UEs dropped are in an active packet session. These packet sessions consist of multiple packet calls representing Web downloads or other similar activities. Each packet call size is modeled by a truncated Pareto distributed random variable producing a mean packet call size of 25 Kbytes. Each packet call is separated by a reading time. The reading time is modeled by a Geometrically distributed random variable with a mean of 5 seconds. The reading time begins when the UE has received the entire packet call.

Each packet call is segmented into individual packets. The time interval between two consecutive packets can be modeled in two ways, as an open loop process or as a closed loop process. The open loop process models the timer interval as a geometrically distributed random variable. Specifically, the mean packet inter-arrival time will be set to the ratio of the maximum packet size divided by the peak link speed. The closed loop model will incorporate the "slow-start" TCP/IP rate control mechanism for pacing packet traffic. Slow-start will be implemented as described in [5]. A total round trip network delay of 100 ms will be assumed for TCP ACK feedback.
The fundamentals of the data-traffic model are captured in Table 6.

Table 10. Data-traffic model parameters.

Process	**Random Variable**	**Parameters**
Packet Calls Size	Pareto with cutoff	? =1.1, k=4.5 Kbytes, m=2 Mbytes, μ = 25 Kbytes
Time Between Packet Calls	Geometric	μ = 5 seconds
Packet Size	Segmented based on MTU size	(e.g. 1500 octets)
Packets per Packet Call	Deterministic	Based on Packet Call Size and Packet MTU
Packet Inter-arrival Time (open- loop)	Geometric	μ = MTU size /peak link speed (e.g. [1500 octets * 8] /2 Mbps = 6 ms)
Packet Inter-arrival Time (closed-loop)	Deterministic	TCP/IP Slow Start (Fixed Network Delay of 100 ms)

12.3.4 UE mobility model

A static or dynamic UE mobility model can be used. Both fixed UE speed or a speed distribution may be used. In the latter case the speed distribution given in Figure 5 shall be used, see also Table 7. A speed is assigned to each user at the beginning of the simulation and will not be changed during the simulation. Stationary UEs signal paths will be Rician faded with K factor of 12dB and 2Hz Doppler spread.

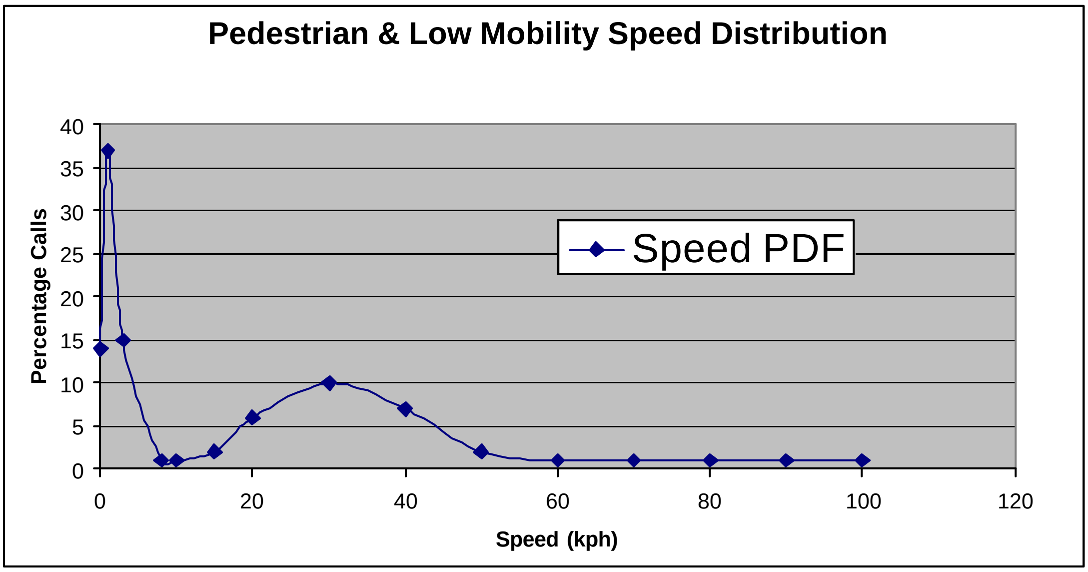

Figure 27. Pedestrian and low mobility speed distribution.

Table 11. Speed distribution

Speed (kph)	0	1	3	8	10	15	20	30	40	50	60	70	80	90	100
Percentage	14	37	15	1	1	2	6	10	7	2	1	1	1	1	1

12.3.5 Packet scheduler

Multiple types of packet schedulers may be simulated. However, initial results may be provided for the two simple schedulers provided below that bound performance. The first scheduler (C/I based) provides maximum system capacity at the expense of fairness, because all frames can be allocated to a single user with good channel conditions. The Round Robin (RR) scheduler provides a more fair sharing of resources (frames) at the expense of a lower system capacity.

Both scheduling methods obey the following rules:

An ideal scheduling interval is assumed and scheduling is performed on a frame by frame basis.

The "frame" is defined by the HSDPA concept, e.g. 0.67ms (1 slot), 3.33ms (5 slots), or 10 ms (15 slots).

A queue is 'non-empty' if it contains at least 1 octet of information.

Packets received in error are explicitly rescheduled after the ARQ feedback delay consistent with the HSDPA definition.

A high priority queue is maintained to expedite the retransmission of failed packet transmission attempts. Entry into the high priority queue will be delayed by a specified time interval (e.g. 5 frame intervals) to allow for scheduler flexibility[4]. If the packet in the high-priority queue is not rescheduled after a second time interval (e.g. 10 frame intervals) it is dropped.

Packets from the low priority queue may only be transmitted after the high-priority queue is empty.

Transmission during a frame cannot be aborted or pre-empted for any reason

The C/I scheduler obeys the following additional rules:

At the scheduling instant, all non-empty source queues are rank ordered by C/I for transmission during a frame.

The scheduler may continue to transfer data to the UE with the highest C/I until the queue of that UE is empty, data arrives for another UE with higher C/I, or a retransmission is scheduled taking higher priority.

Both high and low priority queues are ranked by C/I.

The RR scheduler obeys the following rules:

At the scheduling instant, non-empty source queues are serviced in a round-robin fashion.

All non-empty source queues must be serviced before re-servicing a user.

Therefore, the next frame cannot service the same user as the current frame unless there is only one non-empty source queue.

The scheduler is allowed to group packets from the selected source queue within the frame.

---

[4] The delayed entry into the high priority queue can be used to reduce compulsory retransmission of a single packet. A fast retransmission mechanism, such as N-channel stop-and-wait ARQ, would provide one packet to the high priority queue if the delayed entry mechanism were not provided. As a result, this single packet would be retried in lieu of all other packets regardless of the channel conditions. Note that the case when retransmitted packets always have priority over new transmissions is included in this description as a special case.

12.3.6 Outputs and performance metrics

The following suggested performance metrics for both the entire system and the center site taken over each simulation run may be provided. In all cases, a packet is as defined by the traffic model.

Percentage of users as a function of throughput for different loading levels

Throughput is measured on a per packet basis and is equal to the number of information bits divided by the total transmission time. In other words, retransmissions are accounted for and reduce the peak data rate statistic. The total transmission time is defined to include the time to transmit the initial attempt and each subsequent retry.

For example, consider a packet "*m*":

Packet *m* contains I_m information bits.

Packet requires three attempts to transmit.

Packet m takes $T_{m,j}$ seconds to transmit for attempt j

$$R(m) = \frac{I_m}{\sum_{j=1}^{3} T_{m,j}} \qquad (1)$$

Figure 6 shows a sample output graph.

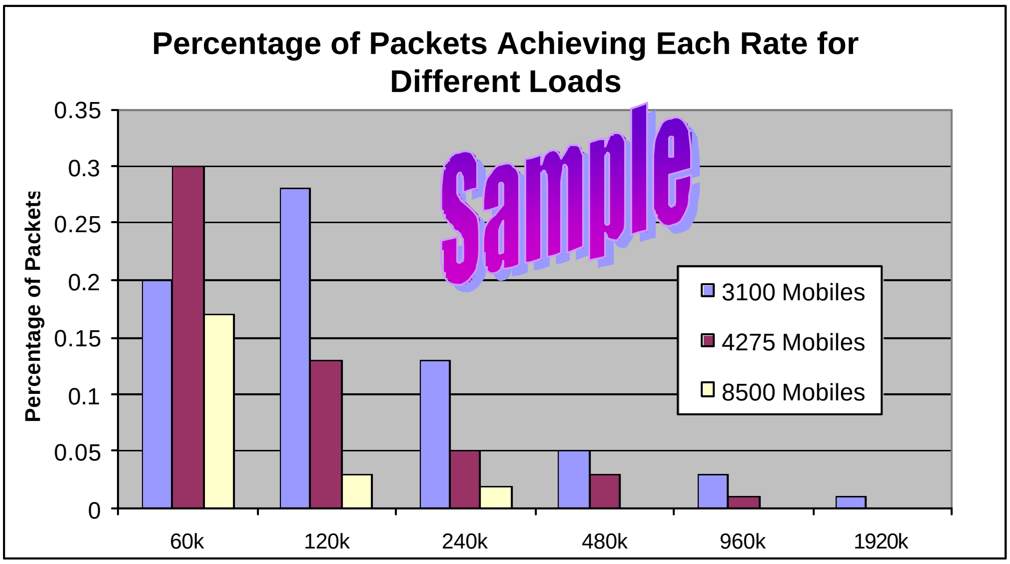

Figure 28. Percentage of Packets as function of throughput for the different loading levels.

Mean distance from serving site for each throughput level, measured per packet

The rate of each packet is calculated as in the previous section. A sample output graph is shown in the Figure 7.

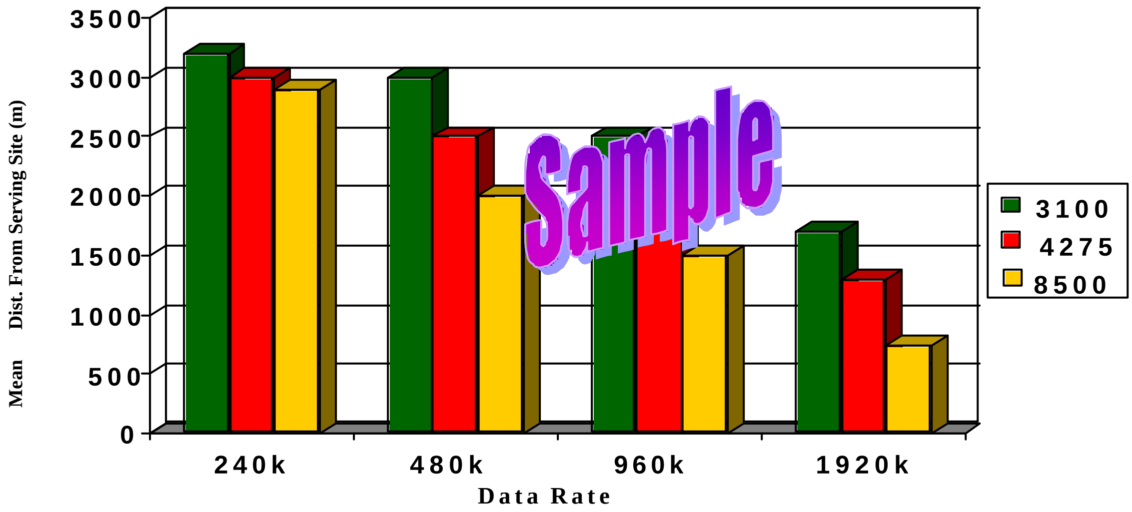

Figure 29. Mean Distance from serving site for each throughput level.

The following statistics as a function of offered load may also be provided

Throughput per sector: Total number of bits successfully transferred divided by the total number of sectors and simulation duration.

Average and Variance of Packet Call Completion Time – measured from when the first packet of a packet call arrives at the base station's queue to when the final packet of the packet call is received by the UE station

Average and Variance Packet Call Transfer Rate - defined as the payload size of a packet call divided by the transfer time where transfer time is measured from when the first packet of a packet call is transmitted by the base station to when the final packet of the packet call is received by the UE station

Service Rate – the number of completed packet calls per second.

12.3.7 Simulation cases

In order to evaluate the performance of the basic features proposed for HSDPA (AMCS, fast HARQ and FCSS), at least the simulation cases described below should be conducted. In both cases the performance reference is the Rel.-99 system.

12.3.7.1 Case 1

In case 1, adaptive modulation and coding (AMCS) and fast HARQ will be modeled.

The following parameters will be used:

MCS may be selected based on CPICH measurement, e.g. RSCP/ISCP, or power control feedback information

MCS update rate: once per 3.33 ms (5 slots)

CPICH measurement transmission delay: 1 frame

Selected MCS applied with 1 frame delay after receiving measurement report

Std. dev. of CPICH measurement error: 0, 3dB

CPICH measurement rate: once per 3.33 ms

CPICH measurement report error rate: 1 %

Frame length for fast HARQ: 3.33 ms

Fast HARQ feedback error rate: 0%, 1% or 4 %.

12.3.7.2 Case 2

In case 2 all the three techniques (fast HARQ, AMC, and FCSS) will be modeled. The parameters are as for case 1, with the addition of:

?? Cell selection rate: once per 3.33 ms

?? Cell selection error rate: 1 %

?? FCSS request transmission and cell selection delay: 2 frames

12.4 System Simulation Results

In case 2 all the three techniques (fast HARQ, AMC, and FCSS) will be modeled. The parameters are as for case 1, with the addition of:

Cell selection rate: once per 3.33 ms

Cell selection error rate: 1 %

FCSS request transmission and cell selection delay: 2 frames

Power allocated to overhead channels (CPICH, PICH, SCH, BCCH, Dedicated): 30%

Maximum power allocated to DSCH: 70%

Maximum number of retries: 15

Cell maximum power: 17 WattsTx Que/Priority Que: 5 frame intervals / 30 frame intervals

Eb/Nt Implementation Loss: 0dB

Std. Dev. of CPICH measurement error: 0

12.4.1 HSDPA Baseline Performance (AMC, HARQ, FCS, Fast Scheduler, 3.33ms frame) vs Release 99 Bound

The packet data throughput for best effort service for is summarized. Table 12 and Table 13 summarize baseline performance for a data only HSDPA system with a Maximum C/I scheduler and a modified ETSI source model [1]. The different throughput metrics presented are defined in Annex B (note the definition of OTA throughput has been modified from [5]). The MCS used were QPSK R=1/2, 16QAM R=1/2, 16QAM R=3/4, and 64QAM R=3/4.

Table 12 **. Baseline HSDPA Throughput Performance vs Load (entire system)**
with Max C/I Scheduler based on Modified ETSI source model and 30% Overhead

Single Rayleigh Ray, 3kph, FRP=0.98 Blk Size=336 bytes Max C/I, Mod. ETSI 30% Overhead AMC, HARQ, FCS

#Users per sector, Max ovsf codes	Average Throughput Statistics Entire System OTA	Service	Packet call	Percent Utilization	Offered Load	User Packet Call Throughput CDF <32k/64k/128k/384k/1M	%UEs with Residual FER >10-2 / >10-4
	(bps)	(bps)	(bps)	(%)	(bps)	(%)	(%)
012ue/sect, 20size32	2,420,614	393,447	1,588,520	16.2	405,329	00 / 00 / 00 / 00 / 21	0.0 / 0.0
037ue/sect, 20size32	1,997,291	1,148,252	1,274,242	56.4	1,181,163	00 / 00 / 00 / 08 / 43	0.0 / 0.0
056ue/sect, 20size32	2,048,895	1,701,205	1,108,079	80.8	1,744,429	00 / 01 / 04 / 22 / 53	0.0 / 0.1
075ue/sect, 20size32	2,341,232	2,167,045	1,041,787	89.5	2,178,208	02 / 05 / 12 / 32 / 58	0.0 / 0.0
100ue/sect, 20size32	2,795,915	2,677,923	1,009,988	92.2	2,704,459	07 / 13 / 22 / 40 / 62	0.0 / 0.4

Single Rayleigh Ray, 3kph, FRP=0.98 Block Size=336 bytes Max C/I, Mod. ETSI 30% Overhead AMC, HARQ, FCS

#Users per sector, Max ovsf codes	Average Throughput - Entire System OTA	Service	Packet call	Percent Utilization	Offered Load	User PktCall thruput cdf <32k/64k/128k/384k/1M	%users with Res. FER >10-2 / 10-4
	(bps)	(bps)	(bps)	(%)	(bps)	(%)	(%)
012ue/sect, 20size32	2,363,084	402,067	1,569,877	17.9	402,628	0/0/0/1/23	0.0/ 0.9
037ue/sect, 20size32	1,976,833	1,130,675	1,228,870	59.9	1,133,783	0/0/1/13/47	0.1/ 7.7
056ue/sect, 20size32	2,091,348	1,660,774	1,075,564	79.4	1,664,775	1/3/8/27/56	0.5/11.2
075ue/sect, 20size32	2,373,411	2,065,421	1,016,820	85.2	2,066,855	4/9/16/36/61	0.6/ 9.7
100ue/sect, 20size32	2,783,349	2,572,591	983,691	89.0	2,579,717	10/16/25/43/64	0.8/ 7.9

Table 13 **Baseline HSDPA Throughput Performance vs Load (center cell)**
with Max C/I Scheduler based on Modified ETSI source model and 30% Overhead

Single Rayleigh Ray, 3kph, FRP=0.98 Blk Size=336 bytes Max C/I, Mod. ETSI 30% Overhead AMC, HARQ, FCS

#Users per sector, Max ovsf codes	Average Throughput Statistics Center Cell OTA	Service	Packet call	Percent Utilization	Offered Load	User Packet Call Throughput CDF <32k/64k/128k/384k/1M	%UEs with Residual FER >10-2 / >10-4
	(bps)	(bps)	(bps)	(%)	(bps)	(%)	(%)
012ue/sect, 20size32	2,118,458	427,956	1,466,449	20.1	454,575	00 / 00 / 00 / 00 / 29	0.0 / 0.0
037ue/sect, 20size32	1,836,760	1,313,060	1,159,096	70.9	1,336,101	00 / 00 / 00 / 10 / 51	0.0 / 0.0
056ue/sect, 20size32	1,877,997	1,867,354	984,997	95.8	1,924,122	00 / 02 / 07 / 30 / 62	0.0 / 0.1
075ue/sect, 20size32	2,205,427	2,277,296	926,064	99.8	2,289,468	03 / 09 / 17 / 41 / 65	0.0 / 0.0
100ue/sect, 20size32	2,708,189	2,812,661	922,184	100.0	2,836,978	12 / 19 / 29 / 48 / 69	0.1 / 0.3

Single Rayleigh Ray, 3kph, FRP=0.98 Block Size=336 bytes Max C/I, Mod. ETSI 30% Overhead AMC, HARQ, FCS

#Users per sector, Max ovsf codes	Average Throughput - Center Cell OTA	Service	Packet call	Percent Utilization	Offered Load	User PktCall throuput cdf <32k/64k/128k/384k/1M	%users with Res. FER >10-2 / 10-4
	(bps)	(bps)	(bps)	(%)	(bps)	(%)	(%)
012ue/sect, 20size32	2,343,668	532,250	1,522,671	23.1	532,365	0/0/0/0/22	0.0/0.0
037ue/sect, 20size32	1,841,815	1,300,848	1,141,774	75.3	1,324,639	0/0/3/13/51	0.0/7.5
056ue/sect, 20size32	1,875,662	1,880,632	952,051	98.5	1,883,993	2/5/13/39/62	1.0/13.4
075ue/sect, 20size32	2,219,979	2,289,244	894,597	99.8	2,295,336	6/11/19/45/68	1.9/12.5
100ue/sect, 20size32	2,695,379	2,813,727	914,466	100.0	2,822,685	14/20/32/52/71	0.9/7.5

From Table 12 above, the Service throughput averaged over all sectors for the Max C/I scheduler is about 2.7Mbit/s at 92% utilization while the OTA throughput is about 2.8Mbit/s. The overall average Packet Call throughput drops from about 1.6Mbit/s to 1.0Mbit/s as the load increases. Fairness is shown in terms of the per user average packet call throughput outage cdf values given in both tables. For example, for the 12 users per sector load 21% of the users achieve an average packet call throughput of between 384kbit/s and 1Mbit/sand 79% of the users in the system achieve better than 1Mbit/s. Residual FER after Hybrid ARQ is given in terms of the percentage of users with packet loss (residual FER) greater than 10^{-2} and 10^{-4}. For the 75 users per sector load from Table 12, about 99.6% of the user's FER after ARQ (residual FER) is less than 10^{-4} and over 99.9% of the users have residual FER less than 10^{-2}. Small residual FER is important to TCP/IP performance.

Table 13 gives center cell only statistics, and shows that the average Service throughput reaches about 2.8Mbit/s at 100% channel utilization. Average Packet Call throughput drops to about 0.9Mbit/s at 100% channel utilization. Note that the service throughput statistic can still improve once 100% channel utilization is reached for a given sector if there are fewer retransmissions. As surrounding sectors reach 100% utilization the uncertainty of other cell interference level is reduced thus reducing AMC errors and resulting in fewer retransmissions.WCDMA Release 99 throughput performance is bounded by the results given in Table 14 and Table 15 below. QPSK modulation with a maximum peak rate of 2Mbit/s was modeled. Fast scheduling, a 3.33ms frame size, and conventional ARQ (no soft combining) were used. A tighter throughput bound is possible by increasing the frame size (TTI=10ms or 20ms) and increasing the scheduling and acknowledgement latency (this was not done for this study). HSDPA Packet Call throughput performance from Table 12 and Table 13 is about twice that of Release 99 throughput bound results in Table 14 and Table 15. For the 56 user/sector load, 47% of the data users have packet call throughput better than 1Mbit/s (see Table 12) while the Release 99 bound case only has 5% of its users better than 1Mbit/s (see Table 14).

Table 14 **QPSK with 2Mbit/s Pk Rate Throughput Performance vs Load (entire system)**

with Max C/I Scheduler based on Modified ETSI source model and 30% Overhead

#Users per sector, Max ovsf codes	Average Throughput - Entire System OTA	Service	Packet call	Percent Utilization	Offered Load	User PktCall thruput cdf <32k/64k/128k/384k/1M	%users with Res. FER >10-2 / 10-4
	(bps)	(bps)	(bps)	(%)	(bps)	(%)	(%)
012ue/sect, 17size32	1,404,140	385,060	837,077	34.8	388,711	0/0/1/8/69	0.0/19.9
037ue/sect, 17size32	1,453,553	1,026,188	611,505	79.0	1,044,331	1/3/9/34/92	1.1/19.9
056ue/sect, 17size32	1,535,192	1,291,871	501,094	86.3	1,353,155	7/13/25/55/95	0.9/10.4
075ue/sect, 17size32	1,562,766	1,386,232	453,189	89.3	1,406,764	17/25/36/59/96	2.3/4.7
100ue/sect, 17size32	na	na	na	na	na	na	na

Table 15 **QPSK with 2Mbit/s Pk Rate Throughput Performance vs Load (center cell)**

with Max C/I Scheduler based on Modified ETSI source model and 30% Overhead

Single Rayleigh Ray, 3kph, FRP=0.98 Block Size=336 bytes Max C/I, Mod. ETSI 30% Overhead QPSK, 2Mbit/s peak rate, noHARQ

#Users per sector, Max ovsf codes	Average Throughput - Center Cell OTA (bps)	Service (bps)	Packet call (bps)	Percent Utilization (%)	Offered Load (bps)	User PktCall thruput cdf <32k/64k/128k/384k/1M (%)	%users with Res. FER >10-2 / 10-4 (%)
012ue/sect, 17size32	1,423,765	547,474	830,334	43.5	548,755	0/0/0/4/69	0.0/20.4
037ue/sect, 17size32	1,450,735	1,266,567	604,962	96.2	1,296,810	0/3/10/37/90	0.0/15.7
056ue/sect, 17size32	1,556,237	1,560,540	464,713	99.9	1,583,330	8/12/31/64/96	2.4/9.1
075ue/sect, 17size32	1,584,900	1,599,731	440,446	100.0	1,657,337	23/32/48/70/96	8.0/8.4
100ue/sect, 17size32	na	na	na	na	na	na	na

Conclusion:

Best effort packet data average sector service throughput for a HSDPA system with 30% overhead using a maximum C/I scheduler was shown to about 2.7Mbit/s based on quasi-static system simulations. A single ray 3kph Rayleigh faded channel was modeled for each user. At this load level up to 38% of the users in the system still achieved a packet call throughput exceeding 1Mbit/s and less than 10% achieved throughput below 32kbit/s (from Table 12). It may also be noted that HSDPA has twice the throughput of the Release 99 WCDMA throughput bound (see Table 14).

12.4.2 Sensitivity to Propagation Exponent

In [6] we show the effect of propagation exponent (rate of increase of path loss with distance) on throughput and it can be seen that the two are strongly correlated[5]. This emphasizes the fact that good isolation between cells is needed to achieve high throughputs.

A similar relationship is obtained for an unfair scheduler.

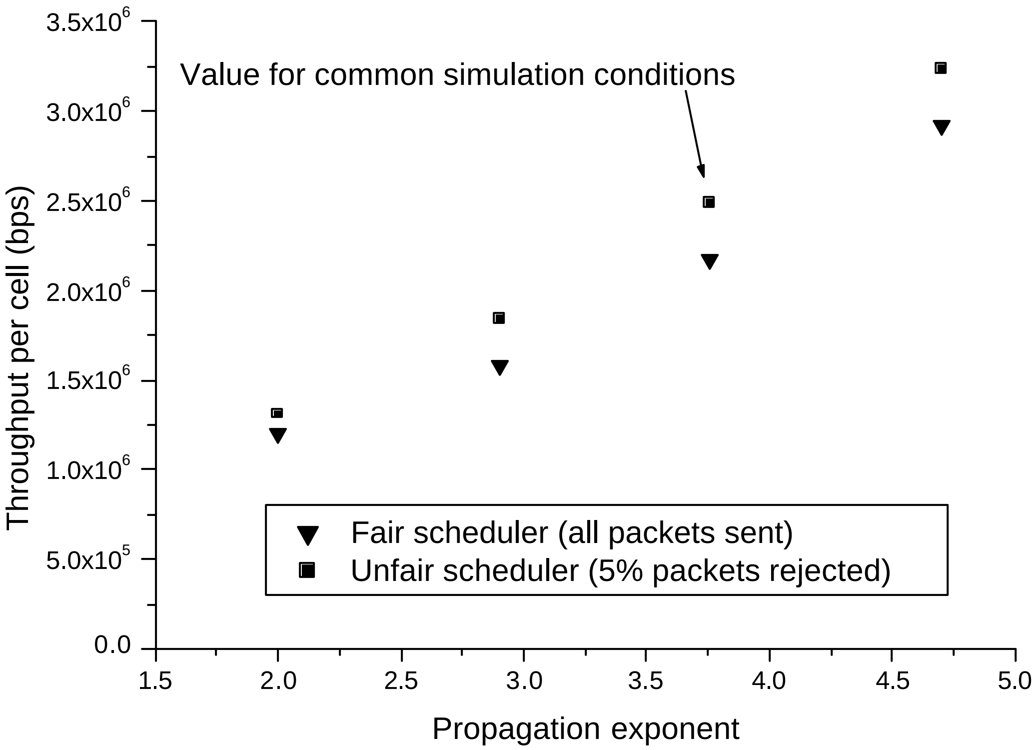

Figure 30 Effect of propagation exponent on throughput

12.4.3 Integrated Voice and Data Performance

Summary:

[5] It may be noted that system simulation assumptions **are not based on Annex A**.

Sector data throughput of an integrated voice and High Speed Downlink Packet Access (HSDPA) system is investigated using the analytical-simulation approach described in [1]. The approach is to integrate the data throughput characteristic (Thruput(x)) in Equation 2 below obtained from link level simulations [2]) for a given channel condition with the achievable carrier to interference ratio (C/I (x)) obtained from system simulation for the coverage area of several representative sectors of the voice & data system.

Equation 2

$$AveSectorThruput = \int_{-20}^{20} Thruput(x) \cdot P(x) \cdot dx$$

The predicted sector throughput is therefore calculated from a combination of link level and system simulations. Figure 31 shows the throughput in bps versus the ratio of energy per chip over other cell interference (Ec/Ioc) at the mobile receiver. We now use Ec/Ioc instead of C/I. Both Hybrid ARQ and transmit diversity (STTD) are enabled. Each curve plotted is in fact a composite of several link simulations for 64QAM R=3/4, 16QAM R=1/2 & R=3/4, and QPSK, using Turbo coding (see [2]). Hybrid ARQ in these simulations uses max-ratio combining of successive attempts (Chase combining). All curves have been simulated at 3kph one-ray Rayleigh fading channel model at a carrier frequency of 2GHz. Link adaptation switches between the modulation and coding levels to maximize the throughput for given Ec/Ioc value.

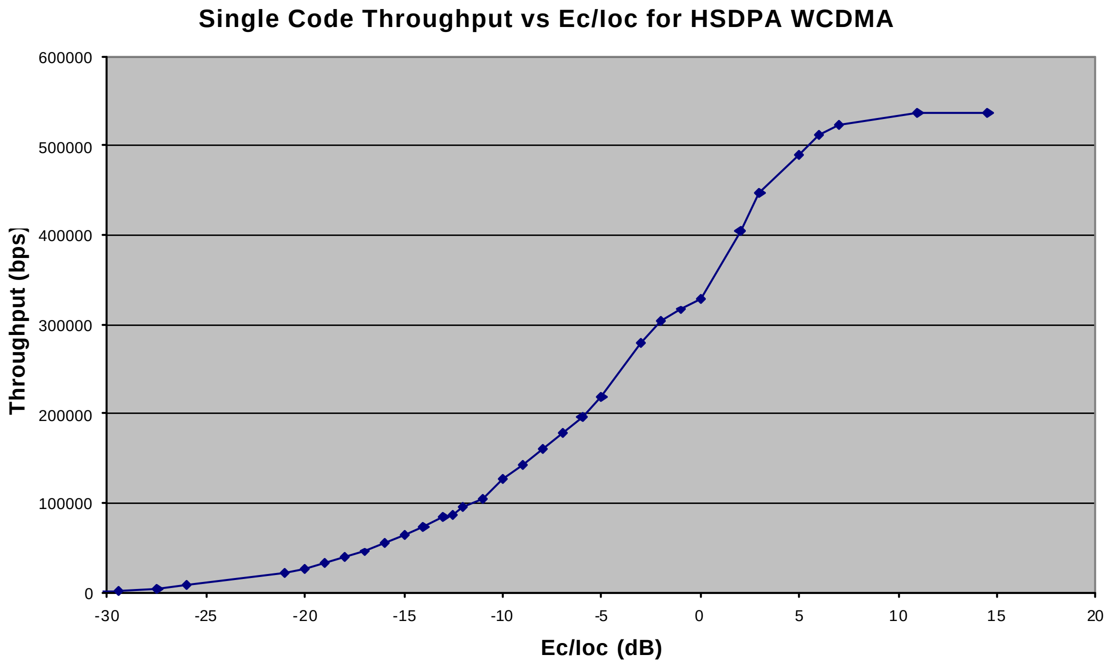

Figure 31. HSDPA Throughput Hull Curve vs Ec/Ioc for 3kph and flat fading

The area probability for a given Ec/Ioc has been calculated from a system simulation of two hexagonal rings comprising 19 3-sector cells with log-normal standard deviation of 8.0dB and a 50% site to site correlation. A full set of radio (Eb/Nt vs FER) curves is used for modeling the 12.2kbit/s voice users. These curves account for 1,2, or 3rays with imbalances from 0 to 12 dB and speeds from 0 to 120kph and geometries (^Ior/Ioc) ranging from –6 to +12dB. The system is assumed to be 100% loaded resulting in the base transceivers having a constant 100% linear power amp (LPA) load of 17 Watts. By always transmitting with constant power (17 Watts in this case (see Figure 35) the voice users will not see abrupt changes in interference levels as the available power margin is allocated to data. Of the LPA load, up to 70% of the power can be allocated to the data channel constructed from up to 20 (or 28) multicodes with spreading factor 32 depending on the voice users (12.2kbit/s) loading. The other 30% of the LPA load is allocated to overhead channels (such as pilot (CPICH), paging (PICH), synchronization (SCH), etc.) and dedicated control channels. The Ec/Ioc area distribution is based on the inner ring sectors and center cell sectors in order to exclude system edge effects .

The sector data throughput for "equal average power" scheduler may be calculated by integrating the throughput from the link simulations against the area pdf for Ec/Ioc derived from the system simulation (see **Equations 2** below). The Ec/Ioc is determined from the available power margin left over after power is allocated to overhead channels (such as pilot (CPICH), paging (PICH), synchronization (SCH)), dedicated control channels, and voice user channels. The number of size 32 OVSF codes, and hence the peak rate that can be allocated, depends on the code tree left over after the overhead and voice channels have been allocated their codes. The equal average power scheduler assigns equal base station power to all users throughout the coverage area achieving the maximum possible throughput for each location. For HSDPA, equal average power scheduling would be achieved by cycling through all users in the coverage area, assigning one 3.33 ms frame with up to 20 (or 28) size 32 OVSF codes and up to 70% of the LPA power while using the optimum modulation and coding level. Over time, each user would receive an equal number of frames and therefore an equal average power allocation from the serving BTS. However, the average data received per user would be biased by the user's location. Users closer to the base site would receive more data then those toward the cell edge.

`Equation 3` (note the power margin could be up to 80% in the sytem simulation. Therefore, in the equations the computed Ec/Ioc is reduced by 0.6 dB to limit the maximum available power for HSDPA data to 70%)

Equation 3

$$\frac{E_c}{I_{or}} ? \frac{P_{m\arg in}}{P_{m\arg in} ? P_{voice} ? P_{ovhd}} ? \frac{P_{m\arg in}}{P_{cell}(j)} \quad \text{for cell } j \text{ at time } t$$

$$\frac{E_c}{I_{oc}} ? 10\log_{10}(\frac{E_c}{I_{or}} ?? \Big/ (I_{on} ? ?)) \qquad I_{on} ? P_{cell}(j)T(j,k) \Big/ \sum_{i?1}^{Ncells} P_{cell}(i)T(i,k)$$

$P_{cell}(j)$ - total power in Watts for cell j (always 17 Watts)

$T(i,k)$ - transmiss ion gain from cell i to probe mobile at location k

? ? fraction of total available power recovered (FRP)

I_{on} ? best serving cell to total power ratio for location k

$$Thrupt ? N_{multi?codes} ?Thruput\_Hull\_Curve(\frac{E_c}{I_{oc}} ? N_{multi?codes\_dB} ? 0.6)$$

<u>Conclusion:</u>

Figure 32 below presents sector throughput of the equal average power scheduler for increasing voice loading and for different (FRP) fractions of total recovered power due to delay spread. For FRP=0.98 and 20 codes the achieved Data only Throughput is about 2.5Mbit/s which then drops almost linearly (see also Figure 33) as voice erlangs per sector increases. For a voice user (12.2kbit/s) load of about 35 erlangs/sector the Data 'equal power' sector throughput is still about 1Mbit/s.

An FRP of 0.98 results in about a 10% loss in throughput relative to an FRP=1.0 while a FRP of 0.92 results in about a 35% loss.

High data sector throughput is maintained by simply allocating the available power margin to data users. This approach is effective as long as the delay from measuring C/I and scheduling for a given user is small. Also the less slots (power control updates) a HSDPA frame encompasses the less margin needs to be set aside for voice users to guarantee them a minimum performance level during a scheduled burst. Alternatively, the power control rate for voice users could be reduced to 500Hz to minimize the power margin needed for voice users over a data frame interval such as 3.33ms.

The draw backs of the kind of simulation-analysis presented are that the effects of voice activity and fast FPC are not adequately modeled and such effects may degrade C/I estimation and hence degrade data throughput. Effects of voice activity and fast FPC will be addressed in later analysis and future quasi-static system simulations. The reverse link will also be modeled.

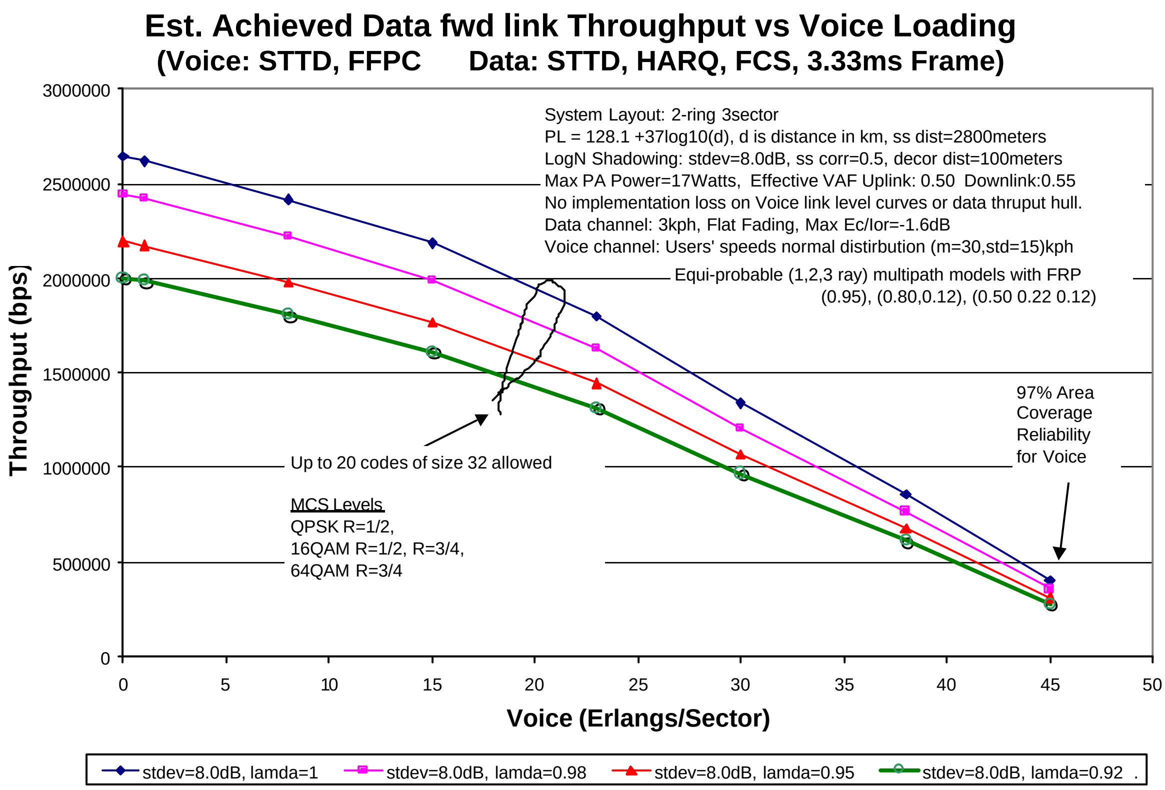

Figure 32 HSDPA Throughput vs Voice Loading for different FRP (lamda) where an Equal Average Power Scheduler is assumed

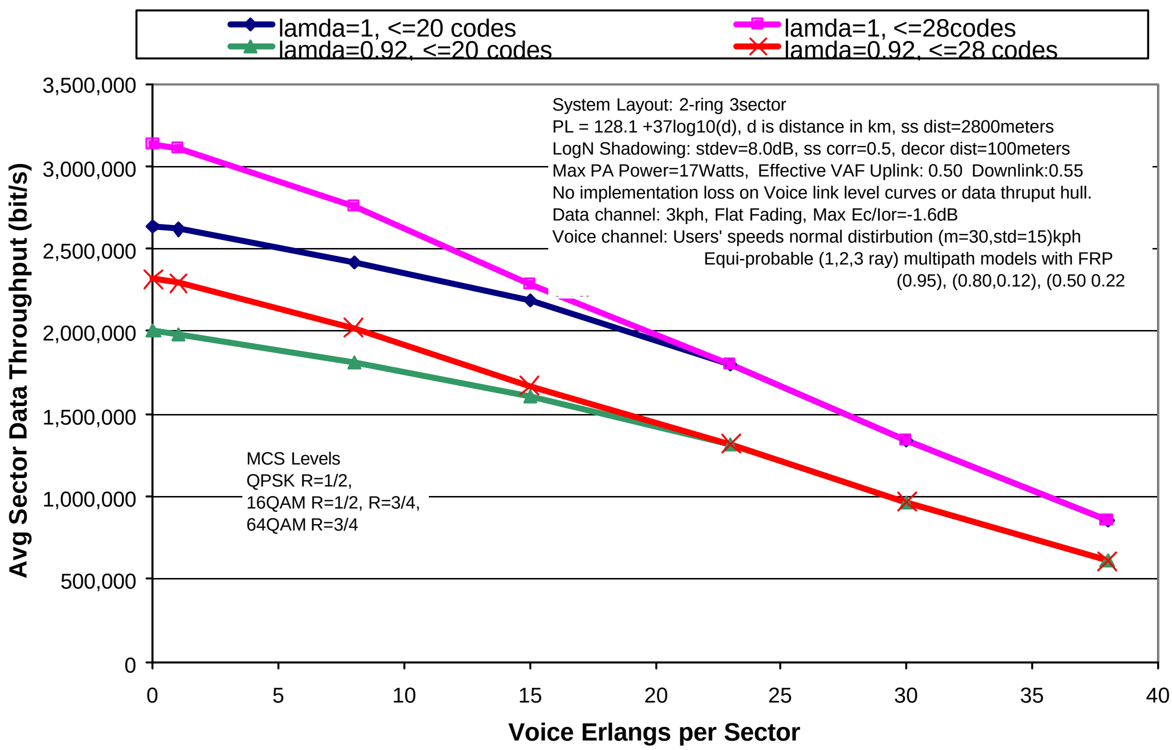

Figure 33 HSDPA Throughput vs. Voice Loading for different FRP (lamda) and for 20 and 28 OVSF codes where and Equal Average Power Scheduler is assumed.

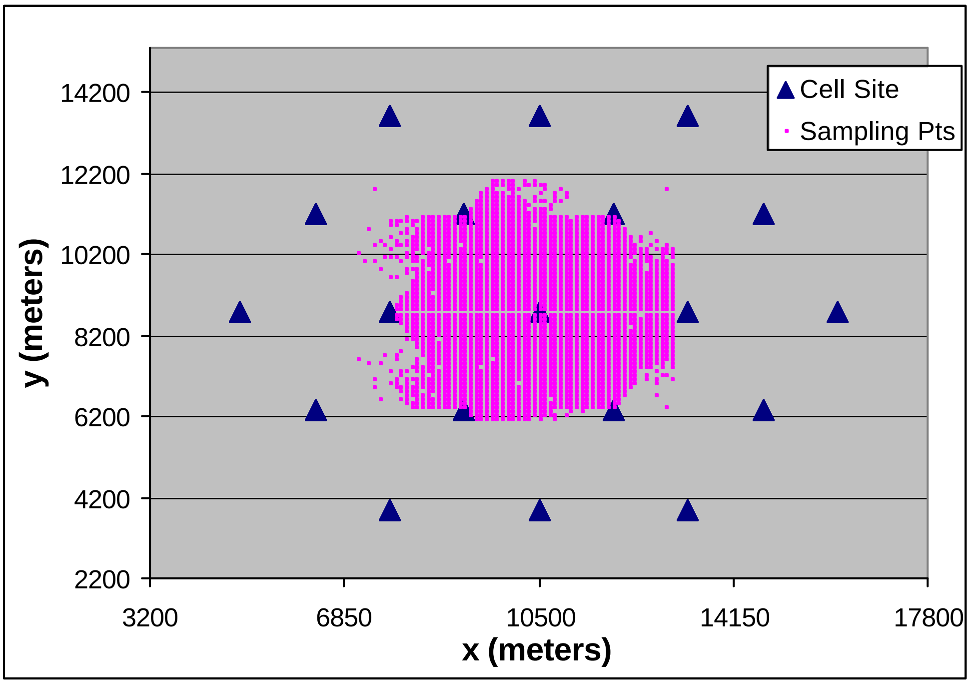

Figure 34 System used for Voice and Data HSDPA Simulation. C/I (Ec/Ioc) sampling points corresponding to coverage area of inner ring sectors and center cell sectors.

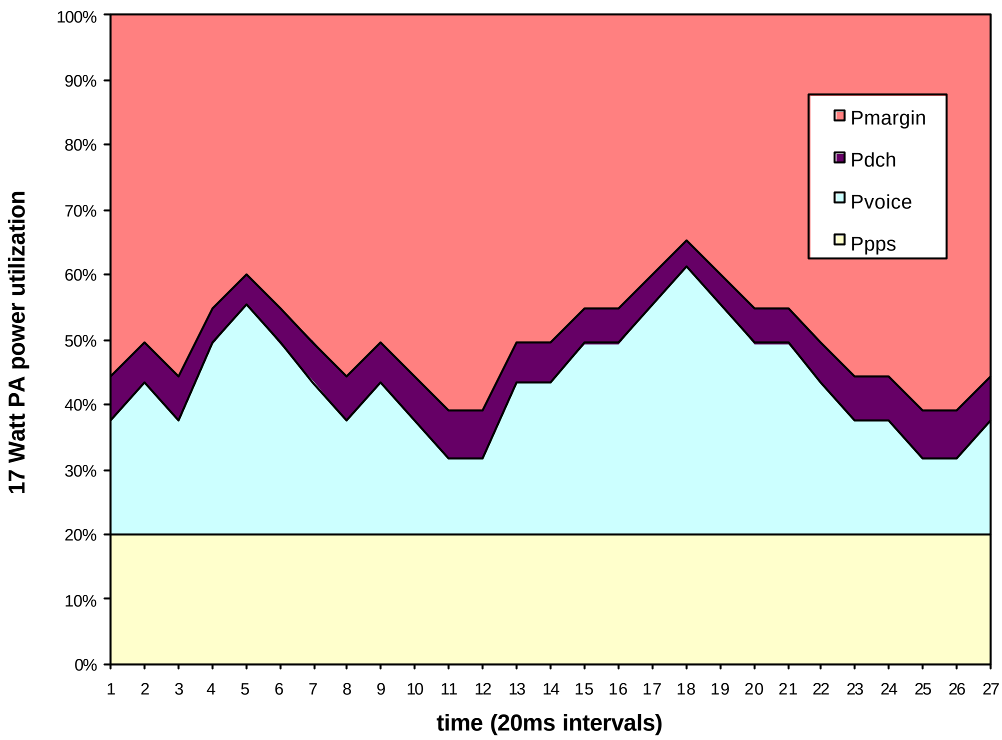

Figure 35 Percentage utilization of 17 Watt PA for a given sector in the system. Note at all times 17 Watts is transmitted for each sector.

12.4.4 Control Information Delay

TPC commands for DL DPCH associated with HS-DSCH may be used to compensate for the DL channel condition report delay and its fundamental ability is evaluated. A concept of the use of TPC commands is illustrated in Figure 36. TPC commands up to T_tx prior to transmission of HS-DSCH TUI are accumulated and used to adjust reported DL channel condition value. More generally, AMCS mode is selected based on DL channel quality estimated as below:

Equation 4

$$DL\_SIR \; ? \; SIR\_report \; ? \; alpha * ? \, TPC * \underset{n}{?} \, (1 \; ? \; 2 * TPC\_cmd[n])$$

SIR_report is reported DL quality reported by UE, $? TPC$ is UL power control step size applied, $TPC\_cmd[n]$ is decoded power control command at slot n, and $alpha$ is adjustment weight factor.

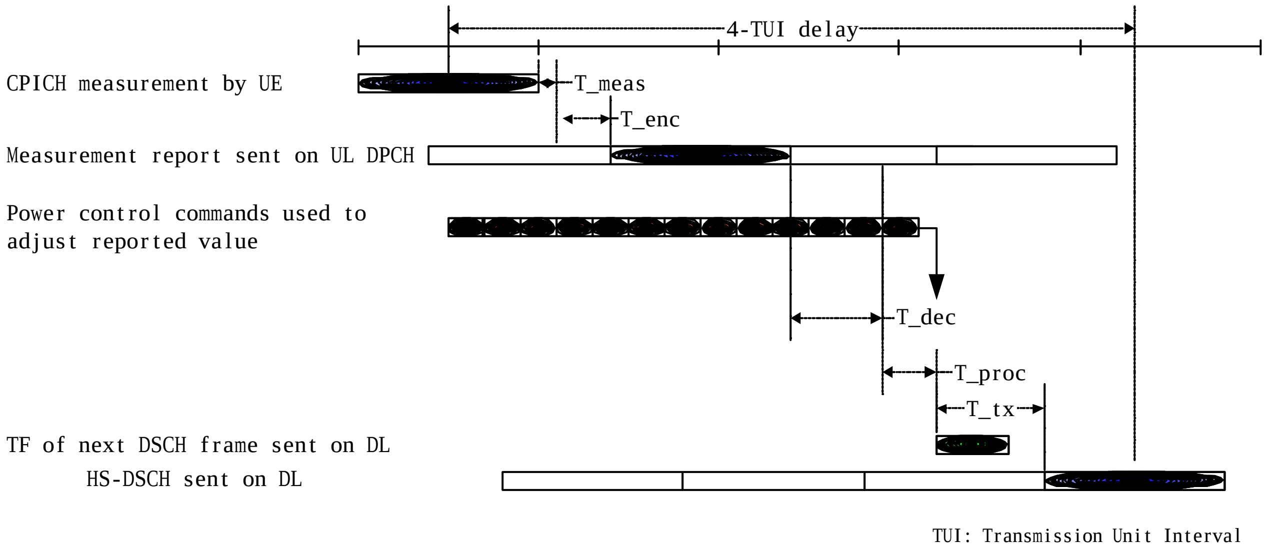

Figure 36 Use of power control commands for DL channel quality report adjustment

Figure 37 and Figure 38 shows the throughput gain achieved using TPC commands to adjust the DL channel quality report. T_tx is set to 4-slot for the evaluation with 3% error in TPC commands. $? TPC$ is set to 1dB and $alpha$ is set to 0.9375 in the simulation. For fading frequency up to 30Hz, gain over the hull characteristics can be achieved.

It must be noted that the gain shown here is dependent on T_tx. From throughput optimization and scheduling point of view, it is desirable to send control information (TF of HS-DSCH) aligned with the HS-DSCH TUI so that T_tx is minimized. On the other hand, for the receiver point of view, it is important that at least channelisation code information be received and decoded prior to receiving the first chip of the HS-DSCH TUI to avoid chip level buffering. Actual gain achievable by TPC adjustment needs to be re-evaluated after a decision is made on TF transmission scheme for HS-DSCH.

It also must be noted that unlike DSCH, DPCH may go into soft-handover state where TPC commands sent from UE does not fully represent DSCH link condition. Further studies need to be carried out to investigate if gain can be achieved in DPCH soft-handover case. Primary cell indicator for DSCH power control, which is in progress for Release 4 work item, may be used to correlate TPC with DSCH link condition .

Conclusions

Longer processing delay of 4 Transmission Unit Intervals (TUI) for DL channel quality report is considered for throughput analysis of HS-DSCH. When considering the control message format and timing for both UL and DL, impact to measurement delay must be kept in mind. It is shown that throughput degradation for 15~30Hz fading condition is significant compared with previous simulation assumption which only allowed 2 TUIs for reporting delay. As for now, a transmission scheme for control messages and expected delay associated with them have not been discussed in detail.

The following issues need to be clarified and studied:

?? DL channel quality transmission scheme (Channel Type)

?? Reporting rate and delay for DL channel quality with respect to HS-DSCH TUI

?? HS-DSCH TF transmission scheme and associated delay

The document also showed some initial simulation results to indicate that use of accumulated TPC commands to adjust reported DL quality is effective to recover some of the throughput loss caused by reporting delay. Further study need to be carried out to validate its gain when associated DPCH is in soft-handover state.

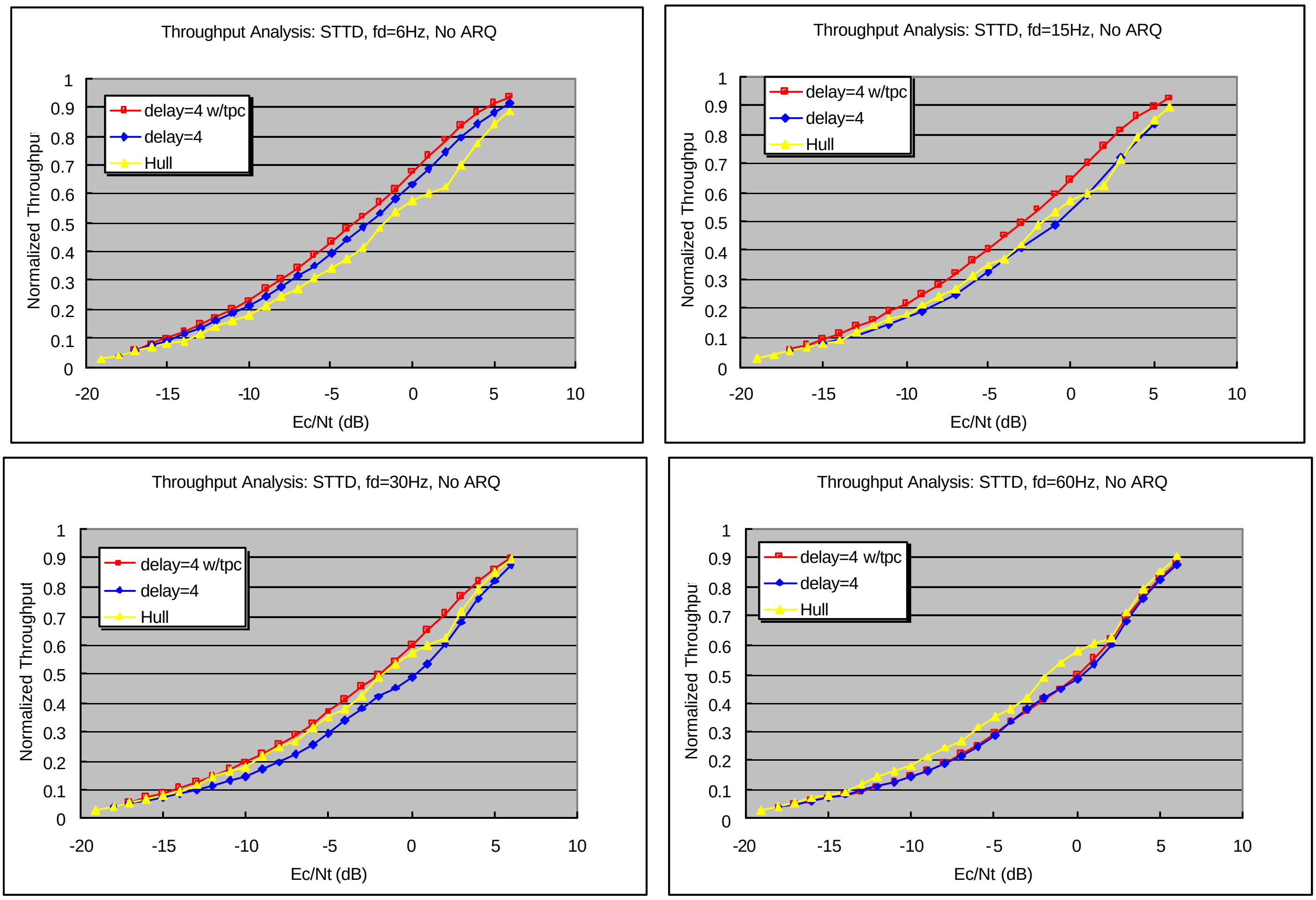

Figure 37 Delay compensation with TPC (No ARQ: TPC error rate=3%)

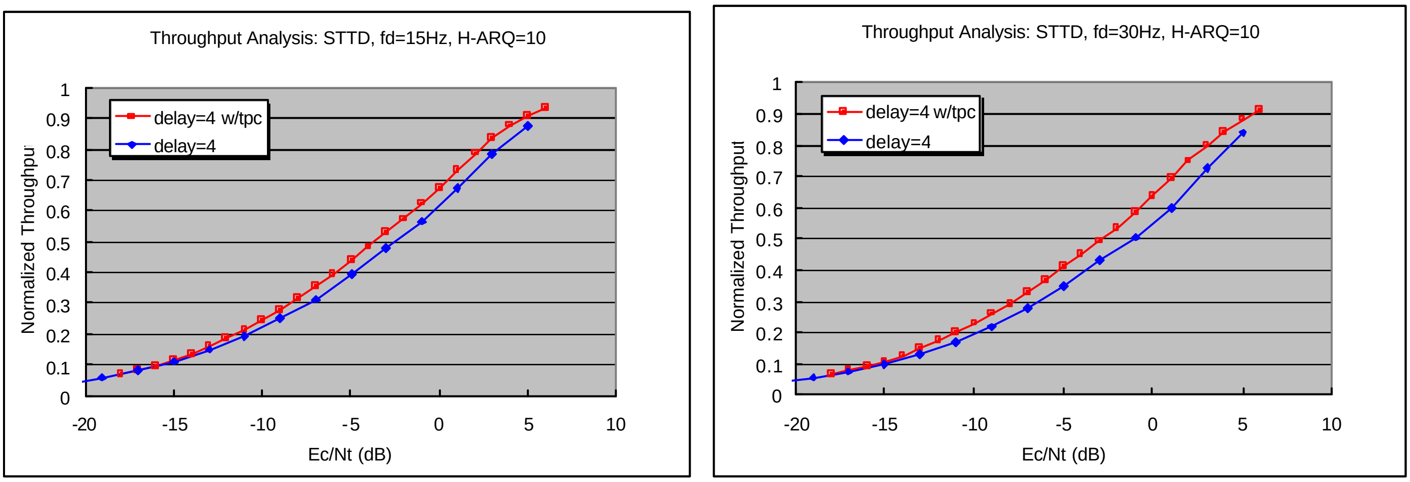

Figure 38 Delay compensation with TPC (ARQ=10: TPC error rate=3%)

References:

[1] Nokia, Ericsson, Motorola. Common HSDPA system simulation assumptions. TSG-R1 document, TSGR#15(00)1094, 22-25th, August, 2000, Berlin, Germany, 12 pp.
[2] Motorola. Evaluation Methods for High Speed Down link Packet Access (HSDPA). TSG-R1 document, TSGR#14(00)0909, 4-7th, July, 2000, Oulu, Finland, 15 pp.
[3] Motorola, TSGR1#17, R1-00-1397.
[4] Motorola, TSGR1#17, R1-00-1398.
[5] Motorola, TSGR1#16, R1-00-1240.
[6] Philips, TSGR1#17, R1-00-1357.
[7] Sony, TSGR1#17, R1-00-1378.

13 Annex B: Examples of Performance Evaluation methods

In the following, two examples of system performance evaluation methods are briefly described. First one is a combination of simulations and analytic evaluation and second one is based on dynamic system level simulations.

A. Analytic Simulation

In this method C/I statistics for all locations in a 19 cell, 3-sectored system is created and the corresponding C/I histogram is obtained. Next, from the link simulations the Throughput vs. C/I results are obtained for various MCS with STTD and Hybrid ARQ. The link and system simulation results are then combined and post-processed to obtain average sector throughput for various classes of scheduler. The flow diagram of this method is shown in Figure 8.

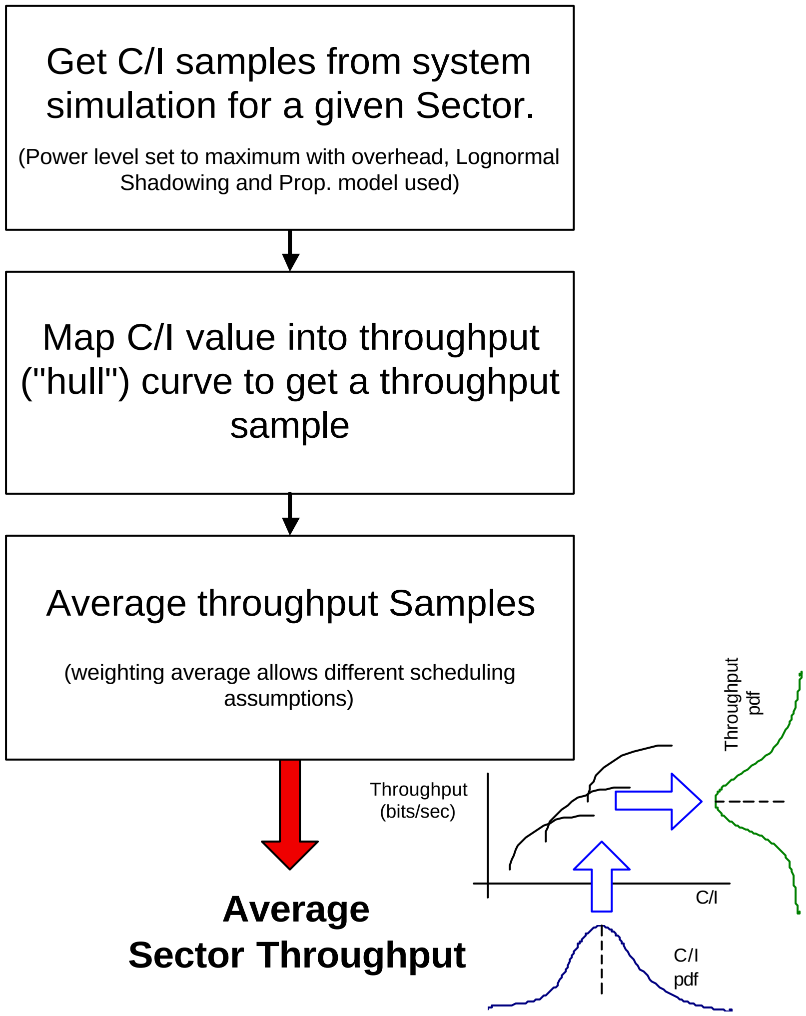

Figure 39. Analytic Simulation Flow Chart

In order to get an estimate of capacity the Max/Min scheduler as shown in Figure 9 is used. In this scheduler users with throughput above their own average get Max/Min more packets than users below the their own average.

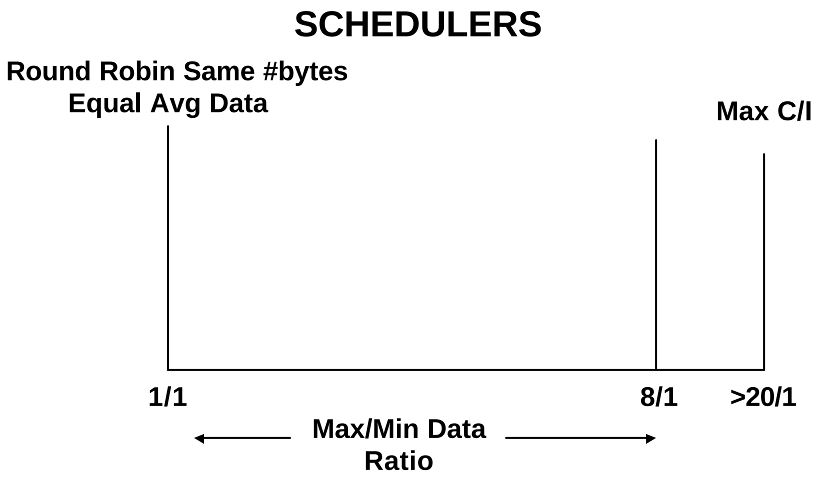

Figure 40. Max/Min schedulers

B. Dynamic system level simulations

Determining high rate packet data system performance requires a dynamic system simulation tool to accurately model feedback loops, signal latency, site selection, protocol execution, random packet arrival, and mobility in a multipath fading environment. The packet system simulation tool will include Rayleigh and Rician fading and evolve in time with discrete steps (e.g. time steps of 0.667ms). The time steps need to be small enough to correctly model feedback loops, latencies, scheduling activities, measurements of required system metrics (e.g. C/I similar to CPICH Ec/No), and fast cell site selection. A Ec/Ior vs. FER curve for a AWGN (static) channel will be created using a link level simulation for each data rate, modulation and coding scheme to determine successful over the air packet delivery. Sampling Eb/Nt points over each frame creates a frame metric. For a given frame the metric is used with the static curve to determine if the frame is erased. Alternatively, one can also use an array of Ec/Ior vs. FER curve for different fading conditions, geometries, speeds and MCS which will then be used in the system simulation to determine whether a frame is erased or not. Lognormal shadowing, delay spread, and fractional recovered power (per ray) will also be modeled. Scheduling and MAC will be included in the simulation to the detail necessary to model resource allocation latencies.

The data traffic model is intended to capture the interaction between radio link algorithms/designs and end-user applications. As such, it is proposed that both best effort and real-time models be simulated to capture air-interface performance. Ideally, best effort services should be modeled by a closed-loop traffic model in the form of a full web browsing model operating over a TCP/IP stack. The close-loop traffic model provides a variable IP packet stream that reacts to the quality of the radio link and the policies of the radio network's packet scheduler. Furthermore, the close-loop traffic model should properly model the bursty nature of data traffic and limit the simulation scheduler to a causal implementation that operates on available information such as the current queue depths and bounds buffering delays to practical levels. The ideal real-time model combines specific frame-erasure rates and delay guarantees to test the capability of the air interface. These real-time models will likely consume greater resources than best effort service. The ability of the air-interface to meet these guarantees may be measured by both the probability of blocking and the residual capacity remaining for best effort services.

14 Annex C: Throughput Metric Definitions

OTA – over the air per frame throughput, Frame Rate/#transmissions. (Unaffected by time between retries.)
Service – total good (successful) frame bits transmitted per second for a given sector. As observed from BTS including all users and idle time. (Affected by time between retries).
Packet Call - total bits per packet call divided by total time to transmit packet call.
Utilization – percentage of time that frame intervals are active for a given sector.
(active = transmission occurs on downlink shared channel).

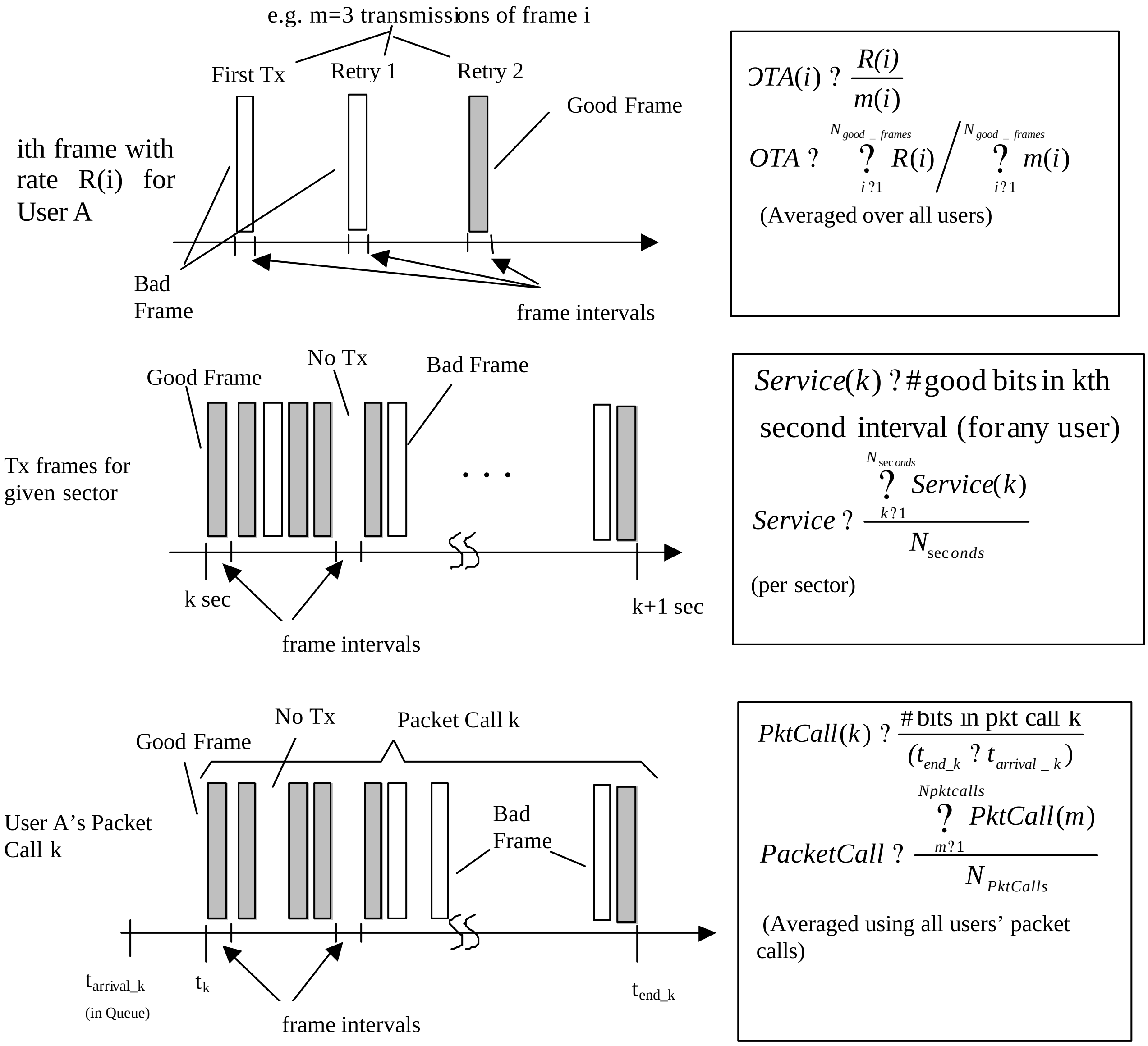

Figure 41 Throughput Statistic Description for System Simulations

The service throughput for a given sector j is

$$ServiceSector(j) = \frac{1}{N_{seconds}} \sum_{k=1}^{N_{seconds}} \#\text{good bits for kth second inverval for sector j} \qquad \textbf{Equation 5}$$

The service throughput averaged over all sectors in the system is

$$ServiceSystem = \frac{1}{N_{sectors}} \sum_{j=1}^{N_{sectors}} ServiceSector(j) \qquad \textbf{Equation 6}$$

Also

$$ServiceSystem = \frac{\text{total good bits all sectors}}{N_{seconds} N_{sectors}} \qquad \textbf{Equation 7}$$

or

$$ServiceSystem = \frac{\text{total good bits all sectors}}{(N_{good\_frames} + N_{retries} + N_{empty}) T_{frame}} \qquad \textbf{Equation 8}$$

where

N_{good_frames} – total good frames over all sectors sent during simulation
$N_{retries}$ – total unsuccessful ("bad") frames over all sectors transmitted during simulation
N_{empty} – total frame intervals over all sectors where there was no transmission during sim.
N_{lost} – total frame intervals over all sectors where the corresponding frame was aborted during sim.
T_{frame} – frame time interval

$$OTASystem = \frac{\textbf{total good bits all users}}{(N_{good\_frames} + N_{retries}) T_{frame}} \qquad \textbf{Equation 9}$$

$$Utilization = \frac{N_{good\_frames} + N_{retries} + N_{lost}}{N_{good\_frames} + N_{retries} + N_{empty} + N_{lost}} \qquad \textbf{Equation 10}$$

$$\frac{ServiceSystem}{OTASystem} = \frac{N_{good\_frames} + N_{retries}}{N_{good\_frames} + N_{retries} + N_{empty}} \qquad \textbf{Equation 11}$$

Therefore

$$\boxed{Utilization = \frac{ServiceSystem}{OTASystem}} \qquad \textbf{Equation 12}$$

The packet call throughput is given by

$$PktCall(k, i, j) \; ? \; \frac{\#\,\text{bits in pkt call k}}{(t_{end\_k} \; ? \; t_{arrival\_k})} \qquad \textbf{Equation 13}$$

where
k = denotes the k^{th} packet call from a group of K packet calls
i = denotes the i^{th} user from a group of N users
j = denotes the j^{th} drop from a group of J drops
the time parameters in Equation 10 are described in Figure A1.
The user packet call throughput becomes

$$UserPktCall(i, j) \; ? \; \frac{1}{K} \overset{K}{\underset{k?1}{?}} PktCall(k, i, j) \qquad \textbf{Equation 14}$$