Exhibit 1015

ZTE Corporation and ZTE (USA) Inc.

◆ Evolution of UMTS Toward High-Speed Downlink Packet Access

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An expanded effort is under way to support the evolution of the Universal Mobile Telecommunications System (UMTS) standard to meet the rapidly developing needs associated with wireless data applications. A new, shared channel—the high-speed downlink shared channel (HS-DSCH)—provides support to packet-switched high-speed data users. A number of performance-enhancing technologies are included in the high-speed downlink packet access (HSDPA) system to ensure high peak and average packet data rates while supporting circuit-switched voice and packet data on the same carrier. Lucent Technologies took a pivotal role in specifying many of these techniques, including adaptive modulation and coding (AMC), hybrid automatic repeat request (HARQ), and fat-pipe scheduling. In this paper, we provide system-level simulations results to indicate the achievable performance and capacity with these advanced technologies. We also discuss HSDPA protocol architecture along with the uplink and downlink control channel design and performance. We conclude with a discussion of potential enhancements for the future. © 2003 Lucent Technologies Inc.

Introduction

The deployment of third-generation (3G) mobile communication systems is under way with support for data rates up to 2 Mb/s (although data rates up to 1.92 Mb/s are possible in Rel99, realistic peak data rate for outdoor environments is about 384 kb/s). These data rates coupled with system latencies will not be sufficient to meet the increasing demands of data services that are anticipated soon after 3G deployment [7]. Therefore, extensive evolution programs in the standards bodies of 3rd Generation Partnership Project (3GPP) and 3rd Generation Partnership Project 2 (3GPP2) were initiated to evolve 3G systems beyond their basic capability in the first release.

Traditionally, voice communication has been the dominant application in wireless networks. As a result, cellular standards, such as Global System for Mobile Communication (GSM) and IS-95, were optimized for voice traffic only. With the recent explosive growth of the Internet, however, a need has arisen to offer both voice and reliable high-speed data access over wireless networks. Until recently, standardized 3G systems such as CDMA2000* and Enhanced General Packet Radio Service (EGPRS) attempted to provide such capability by evolving the air interface of existing voice-centric second-generation (2G) systems. The service needs of voice and packet data, however,

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Panel 1. Abbreviations, Acronyms, and Terms

1xEV-DO—CDMA2000* evolution–data only
1xEV-DV—CDMA2000 evolution–data/voice
2G—second generation
3G—third generation
A^2IR—asynchronous adaptive incremental redundancy
AAIR—asynchronous and adaptive incremental redundancy
ACK—acknowledgment
AMC—adaptive modulation and coding
APP—a posteriori probability
ARQ—automatic repeat request
CQI—channel quality indication
CRC—cyclic redundancy check
CRNC—controlling RNC
DL—downlink
DPCCH—dedicated physical control channel
DPDCH—dedicated physical data channel
DSCH—downlink shared channel
DTX—discontinuous transmission
FCS—fast cell selection
FER—frame error rate
HARQ—hybrid ARQ
HSDPA—high-speed downlink packet access
HS-DPCCH—high-speed dedicated physical control channel
HS-DSCH—high-speed downlink shared channel
HS-SCCH—high-speed shared control channel
IP—Internet protocol
MAC-c—MAC-control
MAC-hs—MAC-high speed
MAC—medium access control
MCS—modulation and coding scheme
MIMO—multiple input, multiple output
MMSE—minimum mean squared error
NACK—negative acknowledgment
NAIR—non-adaptive incremental redundancy
NCCS—network controlled cell selection
RAN—radio access network
RNC—radio network controller
SFN—system frame number
SINR—signal-to-interference-plus-noise ratio
SPARCS—Synergistic Power And Rate Control System
TBS—transport block size
TFRC—transport format and resource control
TF—transport format
TTI—transmission time interval
UL—uplink
UMTS—Universal Mobile Telecommunications System
UTRAN—UMTS terrestrial and radio access network

are different (e.g., low latency and no jitter for isochronous bidirectional streams such as voice contrasted with modest latencies and jitter for packet data, resilience of voice for low frame errors contrasted with extremely low error rates for data applications). Not surprisingly, the support of delay-tolerant data services in these standards proved to be inadequate because voice-centric techniques were applied to do resource allocation for packet data. The recently standardized CDMA2000 1xEV-DO supports efficient packet data service over a dedicated CDMA2000 1X carrier by using a design philosophy that is markedly different from that of CDMA2000 and EGPRS, thereby resulting in a far superior performance. However, 1xEV-DO is not backward compatible with existing 1X systems and, more importantly, does not support voice-service on the same carrier. Hence, an expanded effort is under way in 3GPP and 3GPP2 for the evolution of UMTS and CDMA2000 1X, respectively. These 3G evolutions—high-speed downlink packet access (HSDPA) and 1xEV-DV—address the challenge of supporting the separate and often conflicting needs of voice and high-speed data simultaneously and efficiently on the same carrier in a manner that is fully backward compatible. To address these needs and issues, Lucent Technologies developed its Synergistic Power and Rate Control System (SPARCS). Several key technologies were developed and adapted to address the differing system requirements of HSDPA and 1xEV-DV. This adaptation has required significant system-specific innovation that Lucent spearheaded and successfully championed in the standards bodies. These technologies are described briefly in the section below.

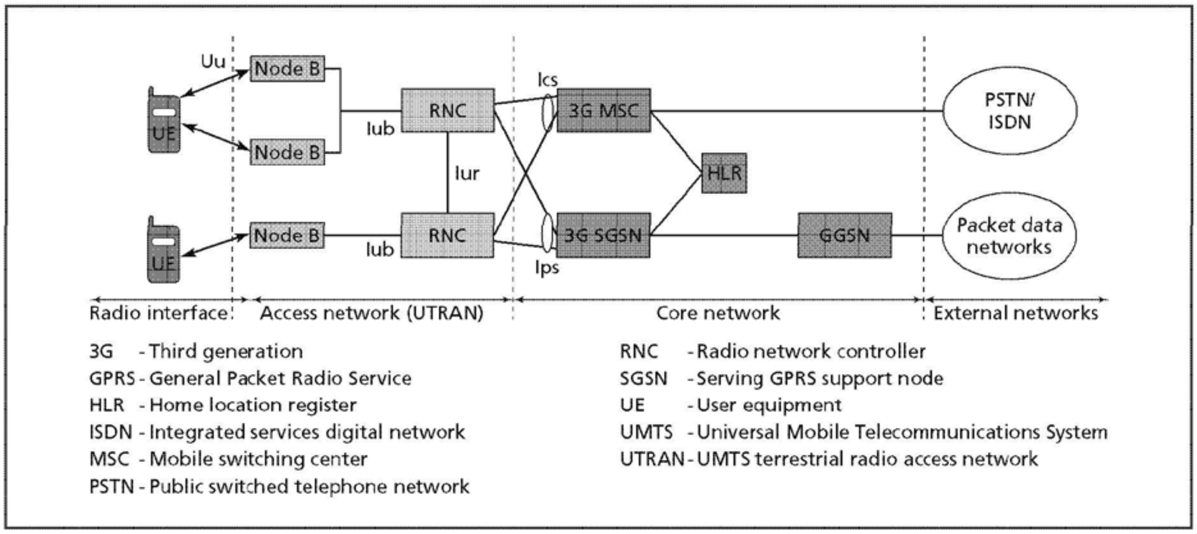


Figure 1.
Simplified UMTS network architecture.

In 3GPP, different areas of evolution of the UMTS system are currently under way. HSDPA is one step in the evolution of UMTS aimed at optimizing the air interface to support higher data rates up to a data rate of 10 Mb/s. Besides the substantial increase in peak data rates, the objectives of HSDPA are to achieve a reduction in system delays and thereby increase system capacity and throughput on the downlink. HSDPA study has shown that, using a single UMTS carrier, a 10.8-Mb/s peak rate is achievable in the downlink, which would significantly increase the downlink packet access speed over the current air interface. This is achieved by implementing a number of new physical layer attributes such as adaptive modulation schemes, fast channel state feedback from user equipment (UE; i.e., mobiles), flexible and dynamic scheduling, and hybrid automatic repeat request (HARQ) channel coding—all within a new suitable architecture. These schemes allow fast link adaptation by selecting appropriate modulation size, number of codes, and the rate of the channel encoder to track variations of the radio channel. While HSDPA is optimized mainly for low mobility urban environments, it will operate well also in other environments with higher mobile speeds. An important consideration in the evolution path of UMTS toward HSDPA was to provide graceful migration for the operators from Rel99/4 to HSDPA capable networks with minimal impacts and costs.

A high-level view of the UMTS architecture is shown in **Figure 1**. It comprises a radio access network part, UMTS terrestrial radio access network (UTRAN), that can interface to a variety of core networks. The core networks contain mobile switching centers and gateways to various circuit and packet networks. UTRAN is linked to the core network via backhaul facilities, for example. T1/E1, STM-x. UTRAN itself comprises cell sites called Node Bs that contain the radio transceivers and radio network controllers (RNCs). Several Node Bs interface with an RNC where, in addition to call setup and control activity, tasks such as radio resource management and frame selection in soft handoff are carried out. Node Bs and RNCs are connected via links that use ATM-based packet transport. UMTS Rel99/4 defines a downlink shared channel that can time multiplex packet data users. However, the scheduling and resource management is performed at the RNC, thereby incurring large delays on the Iub interface. The *downlink shared channel* (DSCH), therefore, is not agile enough to provide very high data rates through dynamic scheduling and rate selection as desired and as outlined here.

For HSDPA, a new channel—the *high-speed downlink shared channel* (HS-DSCH)—is defined and is terminated in the Node B in the UTRAN. This is unlike in Rel99/4 in which the corresponding channel, the DSCH, terminates at the RNC. In addition to enhancements to the air interface, other system capacity enhancements are being developed within both the standards arena and Lucent. Improvements will cover UTRAN and the core network and areas such as distributed scheduler and Internet protocol radio access network (IP-RAN).

Key Features of the HSDPA System

The key technologies used in HSDPA consist of fast scheduling, adaptive modulation and coding (AMC), and HARQ. These technologies along with the advanced channel structure and radio resource control in the Node B (as opposed to the RNC) improve system capacity by a factor of greater than 2 over Rel99 UMTS.

Fast Scheduling

In recent years, considerable work has been done in the area of multi-user transmission on fading channels. Information theoretic results demonstrate that the total downlink information capacity (in the information theoretic sense) is achieved through channel quality sensitive scheduling. In other words, sector throughput is maximized by the scheme in which the base station assigns resources to one user or a subset of users at a given time based on their channel qualities. While traditional forms of diversity in wireless systems include time, frequency, and antennas, such scheduling provides diversity that arises from independent fading channels across different users. Scheduling in conjunction with AMC then allows the base station to select the user with the best channel quality *and* the best-suited modulation and coding scheme (MCS) for that user at the time. Thus, multi-user diversity takes advantage of rather than compensates for channel fading. The sector throughput increases monotonically with the number of users.

Adaptive Modulation and Coding

The benefits of adapting the transmission parameters in a wireless system, especially a CDMA system, to the changing channel conditions are well known. Fast power control is one such example that is critical for CDMA systems with voice users. In the wireless data context, higher data rates can be achieved by varying the modulation level and/or the channel coding rate appropriately based on estimating the channel quality. In a system with AMC, users in favorable positions or users experiencing an "up fade" typically will be assigned higher order modulation and higher code rates. This represents a paradigm shift to rate control rather than power control for wireless data. An added benefit to keeping the power constant is that the inter-cell interference variations on the downlink are reduced.

HARQ

Link adaptation via AMC suffers degradation from a few sources. First, AMC provides limited granularity in data rate selection, and often the channel quality estimates dictate a rate that is in between two allowed MCSs. Second, estimates of link quality are prone to error due to delay between the time of measurement and the time of rate selection and also due to measurement error. HARQ provides some level of robustness through fast retransmissions at the physical layer. Retransmitted copies are combined at the receiver and then decoding is attempted again. The HARQ scheme in HSDPA is based on Lucent's proposal of asynchronous adaptive incremental redundancy (A^2IR) [3]. In A^2IR, the retransmissions can be scheduled exactly like original transmissions. Moreover, retransmissions can use a different number of channelization codes and modulation and coding rates than the original transmission.

Architecture of the HSDPA System

In order to support fast scheduling, adaptive modulation and coding, and HARQ, a new medium access layer called MAC-hs is introduced in the Node B. Moreover, some new control channels both on the downlink and uplink are introduced.

MAC-hs Architecture

UMTS supports a wide range of data rates for services and variable bit-rate operations for a given service. It also allows efficient multiplexing of multiple logical data streams to a user. Many of these activities are accomplished at the medium access control (MAC) layer that resides in the RNC. For example, the MAC layer performs dynamic data rate selection for a service from within a set. This is accomplished by defining a basic unit of time—the transmission time interval (TTI)—and selecting an appropriate transport format (TF) for each TTI. Among other things, the TF determines the number of information bits that will be transferred during the TTI and, thereby, the current data rate for the service. While Rel99/4 supports a shortest TTI of 10 ms, in HSDPA, due to the agility required to exploit fast changing channel conditions, the TTI is shortened to a value of 2 ms.

For HSDPA, the following functionalities have to be added to the existing MAC layer architecture:

- The inclusion of new functionality for HARQ and HSDPA scheduling.
- The provisioning of fast UE channel feedback and fast scheduling or resource allocation that allows the base station (Node B) to take advantage of the good channel conditions of the mobiles (UE).
- A new MAC layer, MAC-hs, located at the Node B in addition to a new channel, the HS-DSCH, that is defined for HSDPA. Relocating the MAC-hs to the Node B facilitates fast scheduling by avoiding the latency involved when the MAC-hs is placed at the RNC.

The entities within the MAC-hs for a UE, illustrated in **Figure 2**, consist of the flow control, the scheduling and priority handling, HARQ, and the transport-format and resource-related information

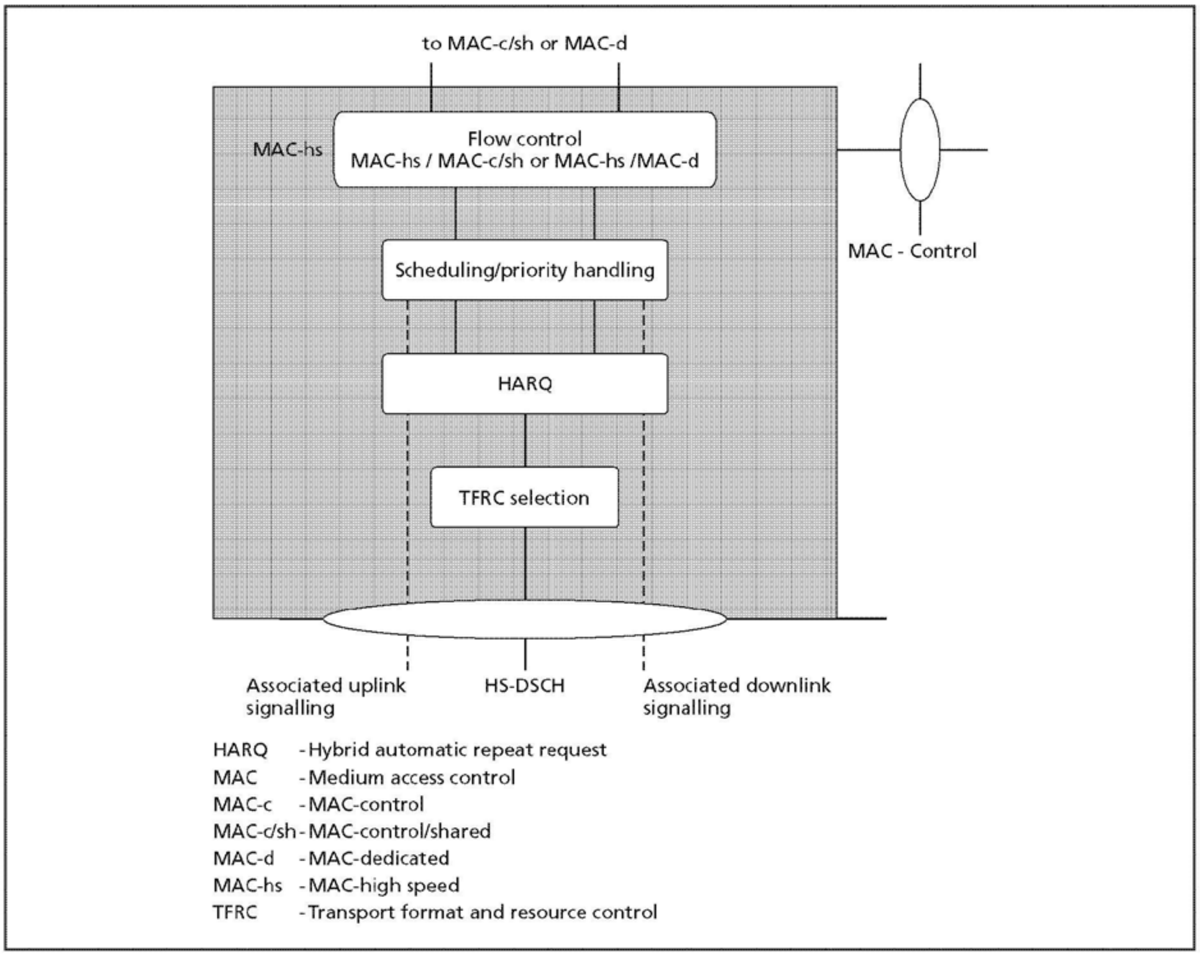


Figure 2.
MAC-hs architecture.

(TFRI) entities. Motivated by the need to perform HARQ operation at the Node B level, the MAC-hs contains the HARQ engine or entity that supports multiple instances of stop-and-wait HARQ processes. At any TTI, the scheduler in the MAC-hs can transmit a maximum of one code block from a single HARQ process from a single UE—i.e., there is only one HARQ process per TTI per UE. With code division multiplexing (CDM), more than one UE can simultaneously receive transmissions in a single TTI.

The data for multiple priority classes is received from the Iub frame protocol at the MAC-hs. The scheduler receives this information from the Iub frame protocol and uses it in making scheduling decisions at the scheduling and priority handling entity. This function manages HS-DSCH resources between HARQ entities and data flows according to their priority class. Based on status reports from associated uplink signaling, it determines whether to send a new transmission or a retransmission. It also sets the priority class identifier and the sequence number for each new data block being serviced.

A flow control mechanism is defined in order to control the amount of data that the RNC can forward to the Node B. This function is intended to limit layer 2 signaling latency and reduce discarded and retransmitted data as a result of HS-DSCH congestion. The flow control is provided independently per priority class.

The last remaining functional entity in the MAC-hs is the transport format and resource control (TFRC) selection entity. Its function is selection of an appropriate transport format and resource combination (i.e., data rate and the physical layer attributes required to achieve it) for the data to be transmitted on HS-DSCH transport channel. The TFRC information is sent in the *downlink high-speed shared control channels* (HS-SCCHs). More than one UE, each with its own TFRCs, can be supported in an HSDPA TTI via CDM.

HSDPA Channel Structure

The HSDPA service is carried over a new channel, HS-DSCH, which is terminated in the Node B. Moreover, a new downlink shared control channel is defined to carry the scheduling and HARQ information to the scheduled UE. In order to provide HARQ feedback information (i.e., acknowledgment/negative acknowledgment [ACK/NACK]) to the Node B, a new channel, the *high-speed dedicated physical control channel* (HS-DPCCH), is defined in the uplink.

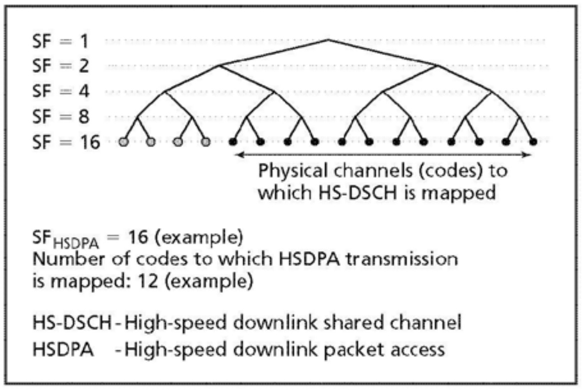


Figure 3.
HSDPA mapping to physical channels with fixed spreading factor.

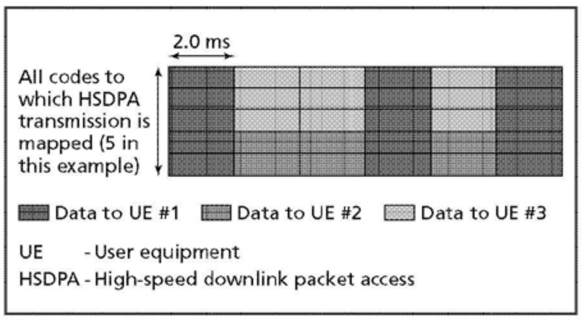


Figure 4.
Sharing by means of time multiplexing as well as code multiplexing.

HS-DSCH. The HS-DSCH preferably supports one UE at a time. However, more than one UE can be code-multiplexed within a TTI if a backlog from the single user cannot fill all the available power and codes within a TTI and/or the UE capability do not allow use of all the available channelization codes. The physical channels to which HS-DSCH is mapped has a fixed spreading factor of 16, as shown in **Figure 3**. The physical channels to which HS-DSCH is mapped can still be shared between "users" in the time domain as well as in the code domain, as shown in **Figure 4**. A physical layer block diagram conceptually showing the transmit chain for this approach is depicted in **Figure 5**.

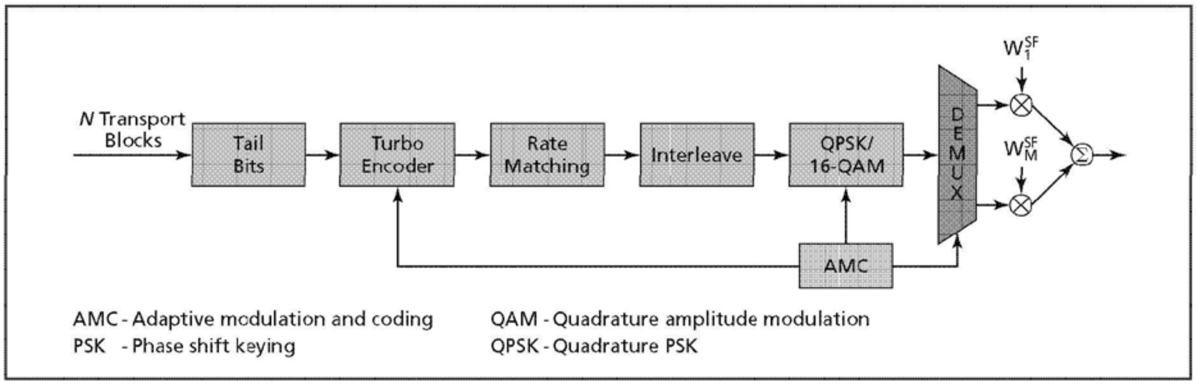


Figure 5.
HSDPA physical layer structure.

HS-SCCH. The adaptive nature of the HS-DSCH provides ample flexibility in radio resource allocation. This flexibility is obtained at the price of fast control signaling. Precisely, for a UE to successfully receive a transmission on the HS-DSCH, the TFRI and the HARQ-related information must be delivered to the UE before the transmission takes place. The detailed control message itself can take up a large amount of downlink bandwidth. Fortunately, due to fat-pipe scheduling of the HS-DSCH, only the few UEs scheduled at a given moment need the control information. Thus, similar to the downlink data packets, the control messages for different UEs are transmitted, when the UEs are scheduled, on a limited number of HS-SCCHs.

For each HS-DSCH, the number of associated HS-SCCHs can range from a minimum of one to a maximum of four. The UE has the capability to simultaneously monitor four HS-SCCHs. For each HS-DSCH TTI, each HS-SCCH carries HS-DSCH-related downlink signaling for one UE. The downlink signaling message contains TFRI (including channelization code set, modulation scheme, and transport-block size) and HARQ information (including HARQ process number, redundancy version, new data indicator, and UE ID). To have the best adaptability to the channel condition, the HS-DSCH transmission should be as early as possible after the channel quality feedback is received. On the other hand, sending the signaling message before the HS-DSCH TTI helps the UE tune to the correct channelization codes and modulation scheme in advance and avoid a huge amount of buffering. As a result, a tradeoff was made to stagger the HS-SCCH and the HS-DSCH (**Figure 6**). Once a UE is scheduled, the signaling message is sent immediately on the HS-SCCH and power controlled toward that UE. The HS-DSCH TTI for the UE is sent two slots after the start of the corresponding HS-SCCH TTI, allowing the UE two slots of time to decode the time critical information, namely, the channelization code set and the modulation scheme used. This not only reduces the buffer requirement of the UE, but also lets the UE start decoding the HS-DSCH earlier.

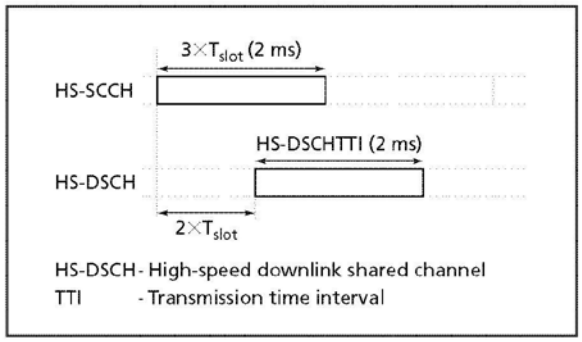


Figure 6.
Timing structure for HS-DSCH control signaling.

The HS-SCCH structure is shown in **Figure 7**. The signaling message is divided into two parts, with part 1 containing the time critical information on channelization code set and modulation scheme, and part 2 consisting of transport block size and HARQ-related

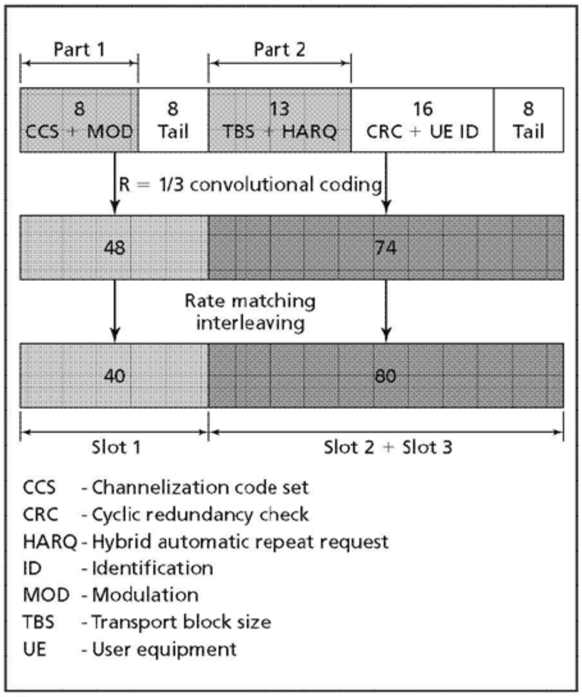


Figure 7.
HS-SCCH structure.

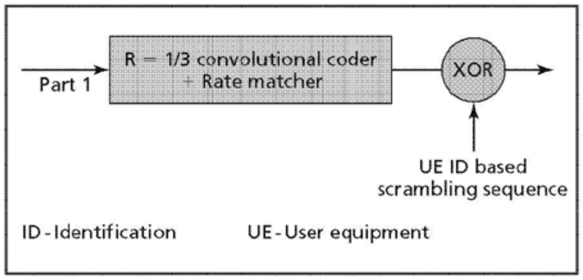


Figure 8.
Scrambling of part 1 by UE-specific ID.

information. A 16-bit UE-specific cyclic redundancy check (CRC) is computed over part 1 and part 2 and attached to part 2. The UE ID is implicitly included in the CRC by using the UE ID as the starting state of the shift register, or using an all-zero starting state, and then masking the computed CRC with the UE ID. These two methods are equivalent. The two parts are further attached with tail bits, convolutionally encoded, and then transmitted in slot 1 and slots 2 and 3, respectively.

At every HS-SCCH TTI, the UE monitors all four HS-SCCHs and tries to extract part 1 information from them. Although there is a UE-specific CRC at the end of part 2 to prevent false alarm for the unintended UE, the UE processing resource is wasted due to buffering of all four HS-SCCHs. This waste in resource is prevented by scrambling the post convolutionally encoded part 1 with UE-specific ID (**Figure 8**). At the receiver, after descrambling by the UE specific ID, a suitable decoder metric (e.g., Viterbi or Yamamoto-Itoh) may be used to determine if the part 1 information was intended for the UE or not. The descrambling, decoding, and validation of part 1 information should be completed before the start of the corresponding HS-DSCH.

As the HS-SCCH carries the signaling message for the HS-DSCH, clearly, successful operation of the HS-DSCH relies on a low HS-SCCH error rate. Unfortunately, unlike the HS-DSCH that can benefit from the AMC and HARQ, the HS-SCCH has a fixed transmission rate and does not allow retransmissions. Although the HS-SCCH is power controlled, its extremely short frame size (2 ms) provides little time diversity. With the presence of fading and channel quality feedback delay, it turns out that the power required to guarantee a certain HS-SCCH frame error rate, say 1%, is quite large. Since the HS-SCCH is sharing power with the HS-DSCH and other dedicated channels, this incurs a significant loss in the system capacity. The power consumption and margin of the HS-SCCH can be reduced by considering transmit and/or receive diversity. Between the two categories of transmit diversity—open-loop and closed-loop—open-loop provides performance gain over the single-antenna system at all mobile speeds. The closed-loop transmit diversity schemes outperform the open-loop schemes at low mobile speed where the feedback rate can track channel variations. At high mobile speed where the feedback loop fails to track the channel variation, the closed-loop schemes provide little gain over the single- antenna system and are no longer better than the open-loop schemes. Considering the additional channel condition feedback required for the closed-loop schemes and the robustness under variable mobile speeds, open-loop transmit diversity schemes are preferred over the closed-loop diversity schemes.

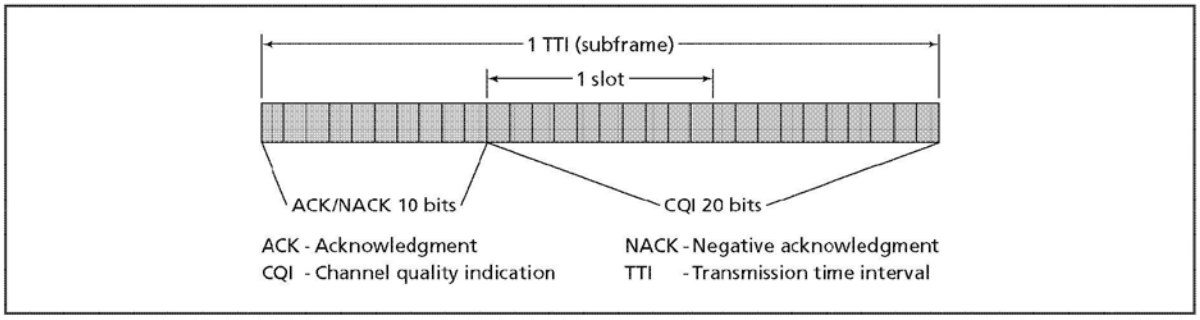


Figure 9.
HS-DPCCH field structure.

The 8 information bits of part 1 and 13 information bits of part 2 are appended with a 16-bit CRC. The generator polynomial for this CRC is chosen to be the same as the one defined in Rel99 UMTS. Additionally, in order to make the information in the HS-SCCH specific to a given UE that is scheduled on the HS-DSCH, the CRC is made UE specific by adding the UE ID modulo two to the 16-bit CRC.

HS-DPCCH. The HS-DPCCH is a new uplink physical channel used by every Release 5-capable UE in order to support HSDPA operation on the downlink. As defined currently, it is used to carry two pieces of feedback information—channel quality indication (CQI) and HARQ ACK/NACK. Physically, it comprises a 256-ary (spreading factor) channelization code framed over three slots (one HSDPA TTI = three slots = 2 ms) comprising two fields for the ACK/NACK and CQI. This channel is code multiplexed along with other uplink physical channels and is carried on either the I or the Q sub-carrier, depending on whether an *uplink dedicated physical data channel* (UL DPDCH) also exists.

Figure 9 schematically illustrates the current working assumption of the structure of the HS-DPCCH sub-frame (or TTI). The ACK/NACK field occupies the first slot worth 10 channel bits, and hence it repeats the one information bit of ACK or NACK ten times. The second and third slots, worth 20 channel bits, are used to carry coded CQI information represented by 5 information bits. A (20, 5) block code whose code words are a linear combination of the 5 basis sequences denoted $M_{i,n}$ defined in **Table I** is used to code the CQI information.

Table I. Basis sequences for (20, 5) code.

I	$M_{i,0}$	$M_{i,1}$	$M_{i,2}$	$M_{i,3}$	$M_{i,4}$
0	1	0	0	0	1
1	0	1	0	0	1
2	1	1	0	0	1
3	0	0	1	0	1
4	1	0	1	0	1
5	0	1	1	0	1
6	1	1	1	0	1
7	0	0	0	1	1
8	1	0	0	1	1
9	0	1	0	1	1
10	1	1	0	1	1
11	0	0	1	1	1
12	1	0	1	1	1
13	0	1	1	1	1
14	1	1	1	1	1
15	0	0	0	0	1
16	0	0	0	0	1
17	0	0	0	0	1
18	0	0	0	0	1
19	0	0	0	1	0

The ACK/NACK bit is used to signal in a fast manner to the Node B by the UE whether the HS-DSCH transmission intended for it was successful or not. Latency is minimized by use of dedicated physical layer resources only for the ACK/NACK creation,

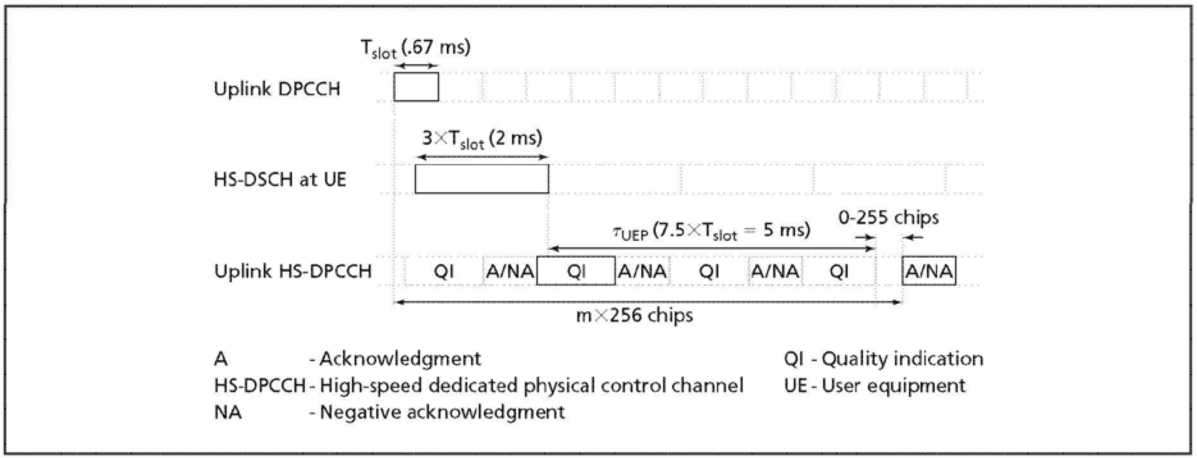


Figure 10.
Timing structure at UE for UL HS-DPCCH control signaling.

turnaround, and processing operations. Latency minimization is critical to minimize the number of HARQ processes spawned for a given UE (that adds to overhead) in order to fully utilize the channel and, more importantly, to minimize the packet delay. This latency is essentially fixed across all UEs regardless of their capability. This leads to a synchronous HSDPA uplink even though downlink scheduling of the HS-DSCH for this UE may be asynchronous for new transmissions as well as retransmissions. Synchronous uplink has the advantage of lower uplink overhead as well as more robustness against misinterpretation by the Node B of the ACK/NACK feedback.

Figure 10 shows the timing offset between the uplink HS-DPCCH and the uplink DPCCH with respect to the downlink HS-DSCH. The code-multiplexed uplink HS-DPCCH starts $m \times 256$ chips after the start of the uplink DPCCH with m selected by the UE such that the ACK/NACK transmission (of duration 1 timeslot) commences within the first 0–255 chips after 7.5 slots following the end of the received HS-DSCH. The UE processing time is therefore maintained at 7.5 slots (5.0 ms) as the offset between DPCCH and HS-DPCCH varies. The ACK bit is sent on the first slot of the code-multiplexed uplink HS-DPCCH. This leaves approximately 4.5 slots-512 chips (propagation delay)-256 chips (HS-DPCCH offset) = 2.8 ms ($T_{Node\text{-}B}$) for Node-B to perform scheduling and signal processing functions with the ACK/NACK bit and (2.5 slots − propagation delay − HS-DPCCH offset) for Node B to perform scheduling and signal processing functions with the CQI information that immediately follows the ACK/NACK bit. The assumption underlying the last sentence is that six TTIs (18 slots) is the minimum separation between retransmissions of the same HARQ process on the HS-DSCH. Of course, the actual separation can be larger depending on the asynchronous scheduler operation for HARQ retransmissions and/or the processing time of the actual Node B implementation. Additionally, the maximum number of HARQ processes possible per UE is eight, due to the three bits overhead allocated on the HS-SCCH for indexing this parameter.

The logic used for ACK/NACK/DTX signaling by the UE is as follows:

- An ACK (0 bit) is signaled if the UE-specific CRC in part 2 of one of the monitored HS-SCCHs passes the check (implying that this UE considered itself scheduled) and the data-specific CRC check passes on the associated HS-PDSCH (implying that the packet was successfully decoded requiring no further retransmissions).
- A NACK (1 bit) is signaled if the UE-specific CRC in part 2 of one of the monitored HS-SCCHs passes the check (implying that this UE considered itself scheduled) and the data-specific CRC

fails on the associated HS-PDSCH (implying that the packet was not yet successfully decoded requiring further retransmissions).

- Neither ACK nor NACK is signaled (the UE shuts off or DTXes the ACK/NACK field) otherwise. This happens when the UE-specific CRC in part 2 of all monitored HS-SCCHs failed the check or such check was not attempted due to the poor metric quality obtained from decoding the UE ID scrambled part 1 information of the HS-SCCH (implying that this UE considered itself not scheduled on any of the HS-PDSCHs). This could also happen when the part 2 CRC check passes for more than one of the monitored HS-SCCHs that is normally impossible, hence suggesting an irresolvable false alarm situation.

The channel quality indicator consists of a recommended TFRC provided by the UE to Node B. When a UE reports a particular CQI, it is reporting that, for the current radio conditions, the UE is able to receive data with a transport format corresponding to the reported CQI, and lower CQIs, at single-transmission PER no greater than 0.1, given the total received HS-PDSCH power of $P_{HSPDSCH} = P_{CPICH} + \Gamma$ (in dB). If S-CPICH is used as reference, the power offset is with respect to the S-CPICH used by the UE; otherwise the P-CPICH is the reference. The power offset Γ is signaled to the UE using higher layer signaling. TFRC reporting as opposed to C/I reporting has been chosen so as to account for the disparity between more and less advanced receivers that for the same observed channel conditions support higher or lower rates.

The CQI for different UE capabilities is determined from TFRC reference tables. For each entry in the table, there is an associated turbo-encoder information word size (expressed as transport block size [TBS] plus 24-bit CRC), modulation type, and number of HS-PDSCH codes. These parameters describe the reference, single-transmission modulation, and coding scheme associated with each CQI value. Note that if—for the current radio conditions and value of Γ—the UE cannot support the minimum CQI (CQI = 1) at PER = 0.1, then the UE reports CQI = 0, which indicates an "out of range" condition to the Node B. To ensure that no UE indicates a transport format exceeding its capabilities, an HS-PDSCH power reduction factor Δ is used to indicate radio conditions beyond the highest transport format. The HS-PDSCH power reduction factor Δ should be interpreted such that the UE is able to receive data with the highest supported transport format assuming $P_{HS\text{-}PDSCH} = P_{CPICH} + \Gamma + \Delta$.

The frequency of the CQI reports as well as the sub-frame offset during which UE transmits CQI can be controlled by higher layer signaling. Lower frequency cycles are intended for use for certain (or all) UEs when they have low activity and/or relatively static channel and/or low capability and/or whenever uplink load on the Node B has to be regulated.

The HS-DPCCH uses certain power offsets (signaled by upper layers from the UTRAN to the UE) that are applied to the ACK, NACK, or CQI transmissions, and these offsets are defined with respect to the power of the UL-DPCCH pilot (Rel99/R-4). Therefore, the HS-DPCCH received signal power is also power controlled in accordance with the Rel/R-4 inner loop power control.

This, however, can cause problems when the UE is in soft handoff and receiving multiple power control commands at the same time from several Node Bs. In the typical situation of "OR of DOWNs," the result is that the UL-DPCCH pilot and the HS-DPCCH received signal power at the HS-DSCH serving Node B (for whom the HS-DPCCH information is destined) can be faded well below normal levels even at low vehicle speeds, in which case the adverse effect is longer lasting. Furthermore, channel estimation based on the UL-DPCCH pilot in order to de-rotate and detect the HS-DPCCH symbols suffers badly. Ensuring that the long-term error rate requirements on the CQI and ACK/NACK channels are met would then require quite large power offsets. This can have a significant impact on the coverage range of the UE (when its transmit power is limited) and on the uplink capacity (due to the injection of a large amount of interference to the HS-DSCH serving Node B as well as surrounding Node Bs).

As a partial solution to the above problem, it was proposed to have UTRAN-controlled repetition factors for the ACK/NACK and CQI fields in order to decrease the power offsets. The repetition factors for these two fields are set independently of each other and range from 1 through 5. The repetition takes place across consecutive sub-frames and, when the ACK/NACK field repetition factor is greater than one, then the UE will not be scheduled for any new or retransmitted data in the corresponding successive HS-DSCH TTIs.

System Capacity and Performance

Traffic and system simulation models and parameters are first outlined followed by some simulation results. The simulation results compare performance of the different HARQ schemes at various speeds and with various schedulers. They indicate that Lucent's proposal of A^2IR, in conjunction with other technologies such scheduling and AMC, considerably outperforms several of the other techniques.

Traffic Model

Performance studies were conducted for a Web-browsing data application. Each user (all users are in active sessions) is assumed to go through a period of Web page download, or packet call, and then spend some amount of time viewing the page before the next packet call. Each packet call comprises packets that collectively represent the objects within the downloaded Web page. The Web page size, the inter-arrival times between packets in a packet call, and the inter-arrival time between packet calls (viewing time) are assumed to be random variables with the distributions outlined in **Table II**. These distributions were agreed upon by the participating companies in the 3GPP standards body and are listed in the HSDPA technical report [1]. An Internet packet size of 1500 bytes is assumed in this study. A modification that was made to reduce simulation time was to reduce the number of UEs required to achieve peak system loading by reducing the average viewing time to 5 seconds (typical values may be 40 seconds or greater).

System Parameters and Assumptions

Basic system-level simulation parameters that were agreed upon in the HSDPA technical report are listed in **Table III**. The following additional assumptions were made in generating the results as follows:

- Seventy percent (70%) of the base station power is available for HSDPA.
- Single path Rayleigh fading channel with 3 km/hr and 30 km/hr speeds is assumed.
- There is no limit on maximum number of HARQ retransmissions.
- Results do not count padding into the throughput (i.e., only information bits count toward throughput).
- Channel quality measurement and ACK/NACK feedback are error free.
- Channel quality feedback delay, which is the time interval between when the UE makes the measurement to when the Node B receives it, is assumed to be 6 slots.
- ACK/NACK delay, which is the time interval from the end of the transmission to the UE to the time

Table II. Traffic model used for system simulations.*

Process	Random Variable	Parameters
Packet calls size	Pareto with cutoff	A = 1.1, k = 4.5 KB, m = 2 MB, μ = 25 KB
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 Mb/s = 6 ms)

*Taken from HSDPA technical report [1].

MTU—Maximum transfer unit

Table III. Basic system level simulation assumptions.

Parameter	Explanation/Assumption	Comments
Cellular layout	Hexagonal grid, 3-sector sites	
Site to Site distance	2800 m	
Antenna pattern	As proposed in [1]	Only horizontal pattern specified
Propagation model	$L = 128.1 + 37.6 \text{ Log}_{10}(R)$	R in kilometers
CPICH power	−10 dB	
Other common channels	−10 dB	
Power allocated to HSDPA transmission, including associated signaling	Max. 70% 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	Unlimited	Retransmissions by fast HARQ
Fast HARQ scheme	Chase combining, non-adaptive or adaptive IR	
BS total Tx power	Up to 44 dBm	
Active set size	3	Maximum size
Frame duration	3 slots or 2 ms, or as indicated	
Scheduling	Max C/I	
Specify fast fading model	Jakes spectrum	Generated, e.g., by Jakes or Filter approach

*Taken from HSDPA technical report.
BS—Base station
CPICH—Common pilot channel
HARQ—Hybrid automatic repeat request
HSDPA—High-speed downlink packet access
Tx—Transmitter
UE—User equipment

at which the Node B receives ACK or NACK, is assumed to be 3 slots.

- The TTI is fixed at 3 slots or 2 ms.
- Parallel transmissions to UEs within a TTI are supported through CDM. However, simultaneous transmissions to a UE on different HARQ channels within a TTI are not allowed.
- Maximum C/I scheduler is used for most comparisons of HARQ schemes. However, a few sample results for other schedulers are provided.

- Neighbor cells are assumed to be transmitting at full power, and HSDPA statistics are collected in the center cell only.
- The notion of fractional recovered power (FRP) is used. FRP is associated with imperfect timing assigned to the RAKE finger at the receiver. FRP represents the fraction of the received energy that is captured at the receiver, and the uncaptured power acts as a source of self-interference. FRP is modeled as an additive white noise source with power that fades with the signal power. The result of such a model for the FRP is to cap the maximum achievable SINR. An FRP of 0.98 is used in these simulations and the resultant maximum achievable C/I to be 16.9 dB.

Data Rates and MCS Selection

The data rate tables used for MCS selection are outlined in **Table IV**. The code block refers to the set of bits that are encoded together using a turbo code of base rate 1/5. All other code rates and sub-blocks are formed by puncturing/repeating the coded bits appropriately.

Although MCS selection may be carried out in a variety of ways with the table, the following approach was used to obtain the results in this paper. Based on the number of codes available and the data backlog, the best MCS that can be supported on the channel at that time is selected. A user's backlog may be rounded up to the nearest code block size via padding or data may be segmented. For the A^2IR scheme, MCS and number of codes may be selected for both first transmission and retransmission. The first transmission of code blocks is always self-decodable ("self-decodable" implies that the code rate for that transmission was smaller than one; non self-decodable transmissions are those for which the code rate exceeds one), but retransmissions are not necessarily self-decodable. If a retransmission corresponds to one of the entries marked "X" in the rate table, then the retransmission is not self-decodable. For such retransmissions, only QPSK modulation is used, and the code rate is selected appropriately.

Performance Metrics

The throughput metrics used are over-the-air (OTA) throughput, service throughput and packet call throughput as defined below.

$$OTA = \frac{\text{Total good bits}}{(\text{Total slots with transmissions}) \times \text{slot duration}}$$

$$\text{Service throughput} = \frac{\text{Total good bits}}{\text{Total slots} \times \text{slot duration}}$$

$$\text{Packet call throughput} = \frac{1}{\text{Num. pkt calls}} \times \sum_{j=1}^{Num\_pkt\_calls} \frac{\text{Bits in pkt call j}}{\text{Duration of pkt call j}}$$

Table IV. Data rate and MCS table.*

Number of Codes	Code Block Sizes, Data Rates, and MCS						
	1280 bits	2560 bits	3840 bits	5120 bits	7680 bits	11520 bits	15360 bits
	640 Kb/s	1280 Kb/s	1920 Kb/s	2560 Kb/s	3840 Kb/s	5760 Kb/s	7680 Kb/s
10	QPSK, 0.13	QPSK, 0.27	QPSK, 0.4	QPSK, 0.53	QPSK, 0.8	16QAM, 0.6	16QAM, 0.8
8	QPSK, 0.17	QPSK, 0.33	QPSK, 0.5	QPSK, 0.67	16QAM, 0.5	16QAM, 0.75	X
6	QPSK, 0.22	QPSK, 0.44	QPSK, 0.67	16QAM, 0.44	16QAM, 0.67	X	X
4	QPSK, 0.33	QPSK, 0.67	16QAM, 0.5	16QAM, 0.67	X	X	X
2	QPSK, 0.67	16QAM, 0.67	X	X	X	X	X

*The cells marked "X" correspond to non self-decodable (code rate > 1) transmissions and may be used only for retransmissions with A^2IR.
MCS—Modulation and coding scheme
QAM—Quadrature amplitude modulation
QPSK—Quadrature phase shift keying

Utilization is defined as

$$\text{Utilization} = \frac{\text{Total slots with transmissions}}{\text{Total slots}}.$$

OTA gives an indication of what bit rates are achievable when transmissions take place, and its dependence on load arises only from the fact that, when the load is heavy and channel-sensitive scheduling is used, the achievable data rate for the selected user is higher on the average. Service throughput is more directly dependent on the offered load as compared to OTA, since unused slots are counted as having zero throughput. Packet call throughput is, loosely speaking, representative of response time for downloads. It is important to note that the packet call throughput as defined above is the average over all packet calls and not an average over users. Therefore, for a Max C/I scheduler (an unfair scheduler) and heavy load, this metric tends to be dominated by the packet calls of good users. Since the packet call throughput metric weights users unequally, it is not representative of fairness or quality of service that users as a whole experience. It is of great importance to additionally consider the packet call throughput cumulative distribution function (CDF) that indicates what fraction of users experience an average packet call throughput greater than the abscissa. Each sample used in determining the CDF is the packet call throughput averaged over an individual user's packet calls. In fact, it is meaningful for operators to define system capacity based on a certain percentile of users achieving a certain average throughput. For example, operators may want to operate the HS-DSCH such that the percentage of users with packet call throughput, below 100 Kb/s, say, is smaller than 5%. The packet call throughput metric averaged over all packet calls can be very misleading in this regard. It is therefore important to compare schemes using the average metrics as well as the CDFs, with the CDFs being the real measure of system capacity.

Simulations Results

In the subsections below, the following schemes are compared. (All schemes are based on stop-and-wait [SAW] with no fixed time relationship between the original transmission and retransmission, i.e., with an asynchronous relationship.)

- Fast physical layer ARQ with no combining (pure link adaptation).
- HARQ with Chase combining [2]. Chase combining results in an improvement in signal-to-noise ratio with each retransmission but not a reduction in code rate. This is because the same parity bits are retransmitted with each retransmission.
- HARQ with non-adaptive incremental redundancy (NAIR). As suggested by the name, this technique is based on incremental redundancy, which results in improved signal-to-noise ratio with each retransmission and, in general, a reduction in code rate with each retransmission. "Non-adaptive" refers to the fact that the modulation scheme has to be the same for retransmissions as in the original transmission.
- HARQ with A^2IR. This scheme, proposed by Lucent and adopted in the 3GPP standards, is based on IR and allows modulation to be changed at the time of retransmission.

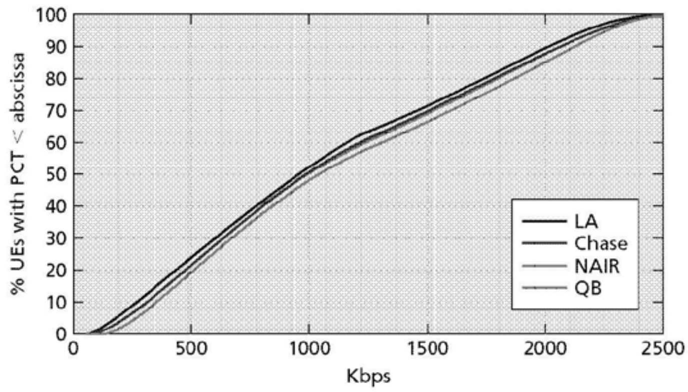
Performance at 3 km/hr. Results from **Figure 11** and **Table V** indicate that at 3 km/hr A^2IR provides the best performance, although the gains from HARQ itself are rather small. This is evident from the fact that the three HARQ schemes—Chase combining, NAIR, and A^2IR—are all quite close to the performance of pure link adaptation. The result is not very surprising because, at 3 km/hr, the channel changes very slowly and AMC provides most of the gains. However, if measurement and quantization errors in CQI are considered, HARQ combining can provide sizeable gains even at slow speeds. Additionally, in some cases when the data rate granularity provided by the rate table is low, A^2IR can provide substantial gains at slow speeds.

Performance at 30 km/hr. At 30 km/hr, the feedback delay (assumed to be 6 slots or 4 ms) between the time of channel quality measurement and the time at which the Node B receives the channel quality report is large enough for the channel to have changed in the interim. It is precisely in such situations that AMC requires the additional robustness to MCS selection provided by HARQ. The user packet call throughput CDFs in

Table V. Average throughput metrics for the different HARQ options.*

HARQ Scheme	OTA (Kb/s)	Packet Call Throughput (Kb/s)	Service Throughput (Kb/s)	Utilization
LA	2621	1108	2561	0.98
Chase	2653	1149	2584	0.97
NAIR	2717	1171	2631	0.97
A^2IR	2704	1200	2587	0.96

*Results are for 75 UEs, 3.0 km/h speed, and a Max C/I scheduler.



Results are for 75 UEs, 3.0 km/h speed and a Max C/I scheduler.

PCT - Packet call throughput
UE - User equipment

Figure 11.
Comparison of packet call throughput CDFs for the different HARQ schemes.

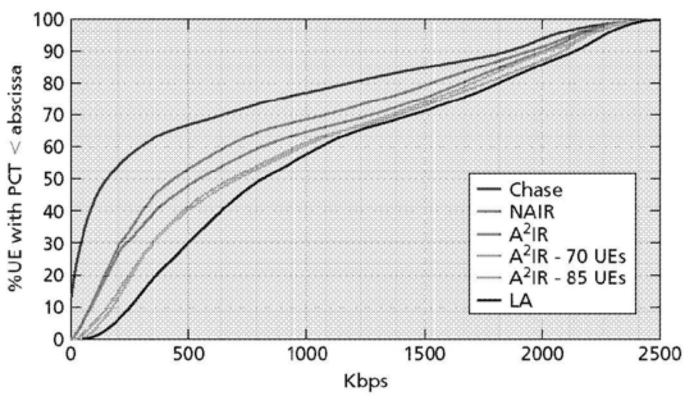
Figure 12 (also see **Table VI**) shows that even Chase combining greatly outperforms pure link adaptation, indicating that in this case HARQ combining is necessary. Furthermore, the two IR schemes, NAIR and A^2IR, perform considerably better than Chase combining, indicating that the coding gain achieved with retransmissions using IR does provide gain. Finally, comparing the two IR schemes, A^2IR outperforms NAIR substantially. This illustrates the fact that adaptivity via rate selection using the residual energy is able to exploit multi-user diversity to a greater degree. If we keep increasing the number of UEs for A^2IR, we find that A^2IR with 70 UEs is close to the performance of NAIR with 56 UEs. Thus, A^2IR yields a gain of at least 33% in system capacity over NAIR. The A^2IR scheme more than doubles the capacity obtained with link adaptation and provides better than 50% improvement in system capacity as compared to Chase combining.

Similar gains are observed when we compare packet call (or service) throughputs for A^2IR with 70 UEs (85 UEs) with NAIR (Chase) 56 UEs. Also note the importance of comparing packet call throughput CDFs because, if we just compare average throughputs for A^2IR with 56 UEs with either Chase combining or NAIR with 56 UEs, the gain is less than 10%. Thus the average throughputs, by the nature of their definitions, hide the true gains obtained with A^2IR over other HARQ combining methods.

Table VI. Average throughput metrics for the different HARQ options.*

HARQ Scheme	OTA (Kb/s)	Packet Call Throughput (Kbps)	Service Throughput (Kb/s)	Utilization
LA	1369	775	1369	1.00
Chase	1756	855	1724	0.98
NAIR	1957	1004	1851	0.95
A^2IR	2129	1075	1960	0.92
A^2IR–70 UEs	2375	982	2310	0.97
A^2IR–85 UEs	2629	929	2621	1.00

*Results are for 56 UEs, 30.0 km/h speed, and a Max C/I scheduler.



Results are for 56 UEs, 30.0 km/h speed and a Max C/I scheduler. For the A^2IR scheme CDFs with 70 and 85 UEs are also shown

CDF - Cumulative distribution function
PCT - Packet call throughput
UE - User equipment

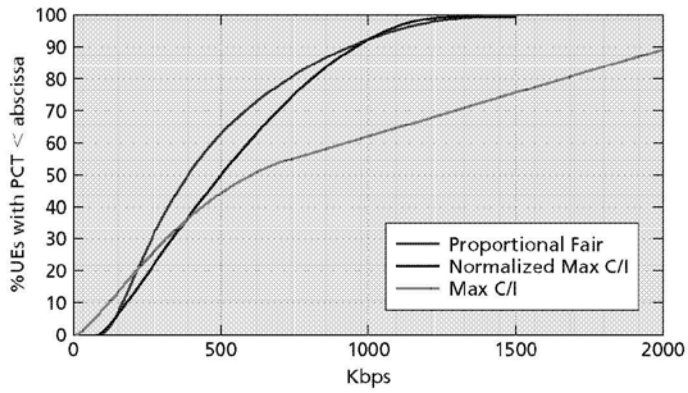
Figure 12.
Comparison of packet call throughput CDFs for the different HARQ schemes.

Scheduler comparison. The scheduler used plays a key role in determining overall achievable system and user throughputs. The Max C/I scheduler achieves high system throughput but is very unfair, as it will pick users at good geometries (i.e., average channel conditions due to location and shadow fading) very often compared to users with poor ones. Additionally, it also deviates somewhat from selecting the user who is at its peak (i.e., experiencing an above its average channel condition). The normalized Max C/I scheduler subtracts (in dB domain) the average geometry experienced by the user and selects the user with the best fade value at a scheduling instant. If all the users have the same fast fading probability distribution function, then this scheme provides fairness in that all users will be scheduled equally on an average (for the infinite backlog case). Furthermore, since each user is picked close to its relative peak, the overall system throughput achieved can also be competitive or even higher as compared to the Max C/I scheduler. Another scheduler that provides fairness without neglecting channel conditions is the well-known

Table VII. Comparison of average throughputs with three schedulers.*

Scheduler	OTA (Kb/s)	Packet Call Throughput (Kb/s)	Service Throughput (Kb/s)	Utilization
Max C/I	2488	984	2449	0.98
Normalized Max C/I	2475	507	2429	0.98
Proportional Fair	2294	564	2261	0.99

*Results are for 30 km/h and 75 UEs.



Results are for 30 km/h and 75 UEs.

PCT - Packet call throughput
UE - User equipment

Figure 13.
Comparison of schedulers.

proportional fair scheduler [10]. In the proportional fair scheduler, an average (moving) of the achieved throughput so-far by each user is maintained. At each scheduling instant, the user with the highest value of the currently achievable bit-rate normalized (directly proportional to channel quality) by the current moving average throughput for the user is selected for transmission. The proportional fair scheduler thus takes both channel quality and previously achieved throughput. A sample comparison of the CDFs for the three schedulers is shown in **Figure 13** and the corresponding average throughputs are in **Table VII**. As expected, because they provide better fairness, normalized Max C/I and proportional fair are better than Max C/I at the lower end of the CDFs and worse at the higher end.

Future HSDPA Enhancements

In addition to all the features and techniques that have been incorporated in the UTRAN specification as described in the "Architecture of the HSDPA System" section, other techniques are being investigated to further improve overall system capacity and performance and user throughput and experience. Some of them are currently under consideration within the 3GPP standards body. Lucent is taking a leadership role in developing these concepts and proposing them for inclusion in future releases. We describe these below.

MIMO Antenna Processing

One of the most promising techniques for significantly increasing data rates and throughputs is to exploit spatial techniques. This has the potential of increasing capacity in a dramatic way. Among the many space–time techniques, BLAST technology [8, 9] utilizing multiple-antenna input at the transmitter and multiple-antenna output at the receiver (MIMO) with code-reuse appears most promising. Fundamental studies have shown that information theoretic capacity for such systems scales approximately linearly with the number of transmit antennas (BLAST systems assume that the number of receive antennas at least matches the number of antennas at the transmitter).

A number of different designs have been proposed to the standards body for inclusion in future enhanced HSDPA systems. Different designs cover different transmission modes (e.g., with or without reusing channelization codes, employing one or per-antenna AMC), receiver modes (e.g., APP or MMSE detection), feedback metrics (e.g., SINR, union bound), and the requisite control channel designs (type and frequency of UL signaling). Solutions expected in the next release of HSDPA will involve a synthesis of the different MIMO techniques, each of which is optimal for a different operating regime or system requirement.

Fast Cell Selection

Using fast cell selection (FCS), the downlink transmission is performed on the best cell in the UE's active set. Thus while multiple cells may be members of the active set, only one of them transmits at any time, potentially decreasing interference and increasing system capacity. Two FCS schemes—intra-Node B FCS and inter-Node B FCS—are considered for HSDPA. In intra-Node B FCS, only the cells within the same Node B can be selected. With inter-Node B FCS, any cell within the active set of the UE can be selected for transmission.

With HSDPA scheduling and HARQ functionality in Node B, there is a need for explicit means to synchronize, e.g., the scheduling and fast HARQ states of the involved Node Bs, in case of fast inter-Node-B cell selection. Different alternatives can be considered for synchronizing the HARQ states: over the RNC by means of network signaling, over the air by means of higher layer signaling, and over the air by means of layer 1 signaling. Over-the-air synchronization by means of physical-layer signaling leads to some undesirable consequences. For example, it is required that this uplink physical layer signaling can be reliably detected by an arbitrary Node B in the active set selected by the UE. 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. On the other hand, synchronizing over the RNC or using higher layer signaling may be more reliable, but cell selection delays will be higher.

In both inter-Node B and intra-Node B, UE selects the cell from which it is willing to receive the transmission. With the UE-controlled approach, the cell selection decision is based on the propagation conditions, i.e., received C/I only. Another approach is to give the network control of the cell selection process. In the network-controlled cell selection (NCCS), determination of the best cell may be based not only on radio propagation conditions, but also on available resources such as power and code space for the cells in the active set. NCCS also allows for load balancing across different cells in order to make best use of the resources. Load balancing is particularly important for bursty data traffic that can create "activity peaks" and "inactivity periods" in different cells at different times.

Multiple Simultaneous Streams

The current HSDPA specification does not allow simultaneous transmissions on different HARQ channels to a UE within the same TTI. The flexibility to support multiple simultaneous transmissions to a UE within a TTI has the potential to enhance HSDPA system performance [6]. The scheduler can use as many resources as necessary for new transmissions and/or retransmissions to the best user in a TTI and use leftover resources for one or more other users. For example, suppose the selected user has a 2,560-bit block pending retransmission on HARQ channel 1 and 1,280 bits of new data in its buffer. If multiple transmissions are allowed, then both the retransmission and transmission of new data can be accomplished in a single TTI. Our preliminary results show an improvement of up to 50% in system capacity at slow speeds when this flexibility is provided. The downlink signaling on the shared control channels remains unchanged from Release 5. The uplink signaling requires multiple ACK/NACKs and this, too, can be readily accommodated with simple changes to the setting and interpretation of the current ACK/NACK field.

Conclusions

HSDPA provides graceful migration for the operators from UMTS Rel99/4 to high-speed data capable networks with minimal impacts and costs. High spectral efficiency is achieved by implementing a number of new physical layer attributes such as adaptive modulation schemes, fast channel state feedback from UEs, flexible and dynamic scheduling, and HARQ channel coding, all within a new suitable architecture. The system capacity achieved with HSDPA is more than twice the UMTS Rel99/4 capacity for packet data traffic. In particular, a system capacity of around 3.0 Mb/s per cell per 5.0 MHz carrier is achieved while still meeting the QoS requirements for data traffic. The further evolution of the HSDPA standard in Release 6 and beyond will include technologies such as MIMO antenna processing, fast cell selection, and multiple stream transmissions to the same user.

*Trademark

CDMA2000 is a trademark of the Telecommunications Industry Association.

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Evolution of UMTS toward high-speed downlink packet access

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Abstract

An expanded effort is under way to support the evolution of the Universal Mobile Telecommunications System (UMTS) standard to meet the rapidly developing needs associated with wireless data applications. A new, shared channel—the high-speed downlink shared channel (HS-DSCH)—provides support to packet-switched high-speed data users. A number of performance-enhancing technologies are included in the high-speed downlink packet access (HSDPA) system to ensure high peak and average packet data rates while supporting circuit-switched voice and packet data on the same carrier. Lucent Technologies took a pivotal role in specifying many of these techniques, including adaptive modulation and coding (AMC), hybrid automatic repeat request (HARQ), and fat-pipe scheduling. In this paper, we provide system-level simulations results to indicate the achievable performance and capacity with these advanced technologies. We also discuss HSDPA protocol architecture along with the uplink and downlink control channel design and performance. We conclude with a discussion of potential enhancements for the future. © 2003 Lucent Technologies Inc.

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