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Wireless Internet Access Using IS-2000 Third Generation System: A Performance and Capacity Study

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Abstract. The Internet has been growing tremendously in the recent years and applications like web browsing are becoming increasingly popular. In a collective effort to provide seamless access to the Internet, wireless equipment manufacturers and service providers are developing 3G wireless systems that efficiently support current and future Internet applications. In this paper, we evaluate the performance and capacity of a 3G wireless data system based on IS-2000 standard. We consider web browsing as the common application for all users and evaluate the system performance for single and parallel web browsing sessions. We perform this study through a detailed simulation of web traffic model described by distributions of number of objects per page, object size, page request size and page reading time. The simulation includes HTTP and TCP/IP protocols, link level recovery, radio resource management, mobility, channel model and, delays in the Internet and the radio access network. We quantify important system attributes like average page download times and system throughput (Kb/s per carrier per sector). We also evaluate normalized object download time, normalized page download time, penalty in performance due to link errors, link layer buffer sizes needed, channel holding time, average power used and distribution of the power used in the system.

Keywords: wireless Internet, HTTP, TCP, RLP, ARQ, capacity, mobility, multiple users

1. Introduction

The Internet offers a wide range of applications starting from interactive web browsing sessions to speech communications to streaming audio and video applications. Traditionally, access to these applications is obtained through wire-line dial-up or LAN connections. More recently, mobile and wireless access systems are being developed to provide untethered access to the Internet. The low data rates (about 10 Kb/s) provided by the existing second generation (2G) wireless systems used predominantly for voice communications (IS-95, TDMA, GSM etc.) are inadequate to provide these wide range of applications. To this end, an extended effort is underway in the wireless and Internet community in standardizing and developing third generation (3G) wireless systems that can provide higher data rates (up to 2 Mb/s). The collective effort of operators and equipment manufacturers has resulted in the development of several 3G standards: in particular, IS-2000 standard [25], UMTS [12] and EGPRS [13]. Air interface standards for these technologies have been completed and systems based on these standards will be deployed in the next several years. Moreover, enhancements to IS-2000 (1x-EV) and UMTS (HSDPA) standards are already underway to support peak data rates of 2 to 20 Mb/s.

As part of the standardization activity, performance analysis has focussed on characterization of the air interface capacity. Since the air interface capacity of a CDMA network is a function of inter-cell interference, the air interface capacity can only be estimated through multi-cell simulations. Such a multi-cell simulation methodology has been specified in [9]. Performance characterization of the physical link in terms of transmit power, intra- and inter-cell interference, and channel characteristics is obtained from link simulations. The results of the link simulations are used in the multi-cell simulations to obtain system "capacity" results. This methodology has been used to optimize the design of the physical layer. The evolution to 3G is motivated primarily by the emerging data applications. It is expected that web browsing will be the predominant high rate data service over wireless networks. Currently, web browsing contributes more than 70% of the traffic carried over the Internet. The web browsing applications employ the Hypertext Transfer Protocol (HTTP) in order to retrieve the information from web servers. HTTP in turn uses Transmission Control Protocol (TCP) as the transport mechanism for reliable data transferring the web pages. TCP is also used by the Internet file transfer protocol (FTP) for reliable data transfer. The performance of TCP over wireless is well known and is extensively studied in the literature [3,4,7,8,17,18,21]. TCP provides end-to-end recovery and flow control. Link layer recovery protocols in cellular systems operate below the TCP layer and have complex interactions with TCP. The link layer recovery protocol can be optimized for best performance with end-to-end TCP. The tradeoffs are different depending on the size of the end-to-end data transfer. A detailed description of this study is found in [19].

While much of the previous studies of HTTP/TCP over wireless have focused on the protocols themselves, this paper provides a comprehensive study of the cumulative effect of these protocols and the lower layers, in the presence of multiple users. We focus on the 3G-1X system which is IS-2000 system with a single radio carrier of 1.25 MHz. We develop a detailed simulation to capture the perceived performance of the mobile Internet users who are randomly scattered in a cell. When there are multiple users in the system the inter-

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action of their TCP sources is complicated by the fact that the radio resources allocated to the different users is varying with time. The base station typically employs a scheduling scheme for determining the bandwidth share for the different users. Scheduling of data users has two immediate effects on TCP flow control. First, the TCP sender sees a varying queuing delay of its data segments. Second, the variability in the data rates allocated to the user, in turn, reflects as variability in the delay experienced by the TCP segments. The subject of scheduling of data traffic is an extensively studied one. The metrics of interest while comparing the scheduling schemes are, typically, the overall throughput achieved by a scheme, the fairness level of a scheme etc. Performance of data users in the presence of TCP flow control and link level recovery is the focus of this paper.

This paper is organized as follows. In section 2 we provide a brief description of the 3G-1X system architecture and elaborate on the key features that affect the performance. In section 3, we provide a description on the functionality of the HTTP and TCP protocols, the link level recovery scheme and the resource access scheme of 3G-1X system. In section 4, we provide a detailed description of the simulation setup used for this study and the important simulation parameters used. In section 5, we provide some of the important simulation results that summarize the performance of 3G-1X system.

2. Architecture and key features of 3G-1X system

In this paper, we study the performance and capacity of 3G-1X CDMA data system. 3G-1X system is the IS-2000 CDMA standard for a 1.25 MHz channel over which a peak air interface data rate of 153.6 Kb/s can be provided. 3G-1X system supports both voice and data. However, since the focus of this work is to study the performance and capacity of 3G-1X system for data applications, we assume no voice traffic.

Figure 1 shows the architectural view of IS-2000 system [23]. For a mobile downloading a web page from the Internet, the data in the web page first arrives at the PDSN. The data is then routed to the PCF, which in-turn sends the data to the SDU. The interface between PDSN and PCF is A10 for user traffic and A11 for signaling information. The interface between PCF and SDU is A8 and A9 for user traffic and signaling traffic respectively. The protocol stack used by different interfaces is shown in figure 1. The signaling interface between Source-BSC and Target-BSC is A7.

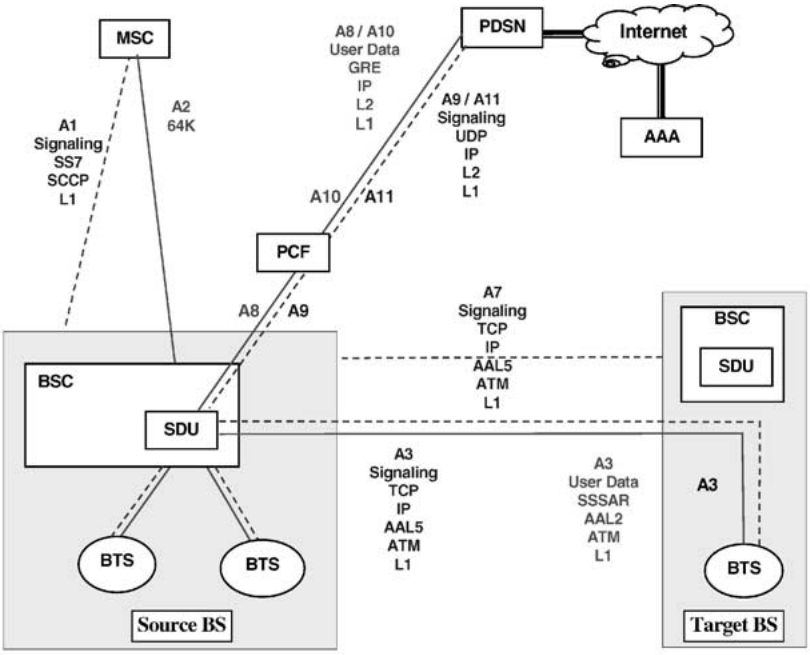
Packet voice, circuit data and circuit voice use a communication path, which is different from that of packet data. A voice call from a mobile is directed to the SDU through the BTS. The SDU then connects the call to the MSC, which connects to the IWF. The IWF connects to the PSTN. The primary functionality of the IWF is to change the speech coding in the cellular phone calls to the regular 64 Kb/s PCM coding of land- line-calls.

CDMA-2000 (IS-2000) provides high data rates by providing the users with a supplementary code channel. The decision of which rate to allocate and when to allocate it is taken by a fairly sophisticated scheduler located in the BSC. The scheduler takes into account the loading in the system, the channel condition of the user, the user's data backlog, priority etc. SDU is responsible for adding the sequence number to RLP frames and A3 interface transports these RLP frames to the BTS. Currently A3 interface between SDU and BTS is AAL2/ATM. It is important to note that as shown in figure 1, A3 interface is defined between Source-SDU and Target-BTS of the system. Hence, this interface comes into the picture only during a soft-handoff scenario. The interface between Source-SDU and Source-BTS or Target-SDU and Target-BTS is not defined by the IS-2000 standard and is deemed proprietary of the manufacturing company.

Given a physical layer data rate, there are specific RLP frame formats that the SDU has to use. In addition to the RLP frame formats, there is a predefined set of Multiplex sub-layer options to be used. The Multiplex sub-layer is a protocol layer situated between Layer 2 and Layer 1. This layer is responsible for multiplexing Layer 2 SDUs from multiple streams from the same user onto the same physical channel, according to priority and QoS criteria.

The data rate that can be allocated to a particular mobile in the cell coverage area is a function of the in-cell and out-of-cell interference at the mobile. Therefore, a rate lower than 153.6 Kb/s (e.g., 76.8 Kb/s or 38.4 Kb/s) may be the peak rate available at certain locations within the cellular coverage area. In order to be backward compatible with existing IS-95 based 2G systems [16] the IS-2000 radio interface retains many of the attributes of the IS-95 air interface design. The IS-2000 system radio interface also incorporates additional features that add more flexibility to the system and improve the performance.

IS-2000 classifies the air-link packet data channels as dedicated (FCH) or shared (SCH). Every data user admitted into the system is first given a dedicated low rate channel. This is called Fundamental Channel (FCH). The peak air-link rate possible on FCH is 9.6 Kb/s. If a user having FCH has enough data backlog, then the user is qualified for an additional Walsh code which works as a Supplemental Channel (SCH). All users qualified for a data rate higher than 9.6 Kb/s are assigned priorities using factors such as their channel quality, data backlog, priority class etc. Then the user with highest priority is assigned a Walsh code, if the system has any remaining, and the peak rate user can achieve is computed using the data backlog, channel condition and available power resources etc. If the system is heavily loaded, resulting in small or no power available to the highest priority user, the user is denied SCH. This process is repeated for the each user in the order of their priority as long as the power resource is available. Once a high data rate is allocated, the amount of time a user can use that rate is dependent of the service/priority class a user belongs to. IS-2000 system allows for 14 such priority classes for users subscribed to *non-assured QoS* and *assured QoS* modes [27] that wireless service providers could use to promote their services. In addition, users in assured mode can communicate parameters



A1	A1 interface carries signalling information between the Call Control (CC) and Mobility Management (MM) functions of the MSC and the call control component of the BS (BSC).
A2	A2 interface carries 64/56 kbit/s PCM information (voice/data) between the Switch component of the MSC and one of the following: Channel element component of the BS (in the case of an analog air interface), and SDU function (in the case of a voice call over a digital air interface)
A3	A3 interface carries coded user information (voice/data) and signalling information between the SDU function and the channel element component of the BS (BTS)
A5	A5 interface carries a full duplex stream of bytes between the IWF and SDU
A7	A7 interface carries signalling information between a source BS and target BS
A8	A8 interface carries user traffic between the SDU and the PCF.
A9	A9 interface carries signalling information between the SDU and the PCF.
A10	A10 interface carries user traffic between the PCF and the PDSN.
A11	A11 interface carries signaling information between the PCF and the PDSN.

Figure 1. Architectural view of IS-2000 system.

such as minimum requested data rate, minimum acceptable data rate, requested data loss rate (loss rate is measured after ARQ by RLP), acceptable data loss rate etc., for the forward and reverse directions separately, during connection establishment. In this paper, we assume that all users belong to the non-assured QoS mode and that they are all subscribed to the same priority level.

3. Description of protocols

3.1. Functionality of HTTP protocol

Hyper Text Transport Protocol (HTTP) is the *de facto* protocol for information retrieval on the World Wide Web. The clients requesting information from a web server follow a series of request/response type interactive messages as defined in the HTTP protocol. Although the most widely used version of HTTP protocol was HTTP/1.0, a modified version HTTP/1.1 is also widely used. A detailed description of these protocols is provided in [5,15].

We consider HTTP version 1.0 in our analysis. In HTTP/1.0 [5], without persistent TCP connection support, a TCP connection is needed for every object in the page. In HTTP/1.0 with persistent connections, a TCP connection is kept open for a short period of time after the object transfer. Therefore, the same TCP connection can be used to retrieve multiple web objects. The persistent connections reduce the page transfer delays by reducing the number of TCP three-way handshakes and slow starts needed for page retrieval. On the other hand, in order to make efficient use of the persistent connections, the number of parallel TCP connections has to be limited to a low value which might in turn contribute to increased page transfer delays.

An example of a web page retrieval using HTTP/1.0 is depicted in figure 2. The web page shown consists of six objects referred to as object A, B, C, D, E and F. In HTTP/1.0, with-

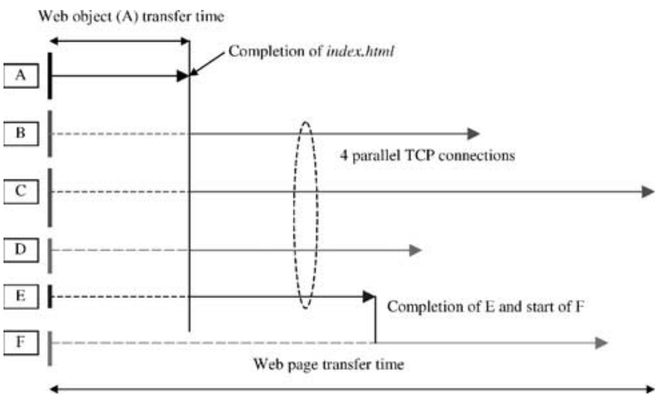


Figure 2. Graphical description of web page transfer using HTTP/1.0.

out persistent TCP connection, a new TCP connection is set up for every object in the web page. Most web browsers set an upper limit on the maximum number of simultaneous TCP connections (C) allowed during a page download. In this example, we assume that the maximum number of parallel TCP connections is limited to 4. The page transfer starts with a single TCP connection for the main object (`index.html`). Once the first object transfer is complete, the client can send GET requests for multiple objects and the parallel TCP connections can be opened.

3.2. Functionality of TCP protocol

The Transmission Control Protocol is a very well developed and popular transport protocol. It is a connection-oriented transport layer protocol, which provides reliable transfer of data from a host to a client. One of the essential features is in-sequence delivery of data by reordering out-of-sequence data and discarding duplicate data. TCP handles the problem of congestion in networks through flow control and the nature of flow control employed distinguishes the several versions of TCP, like Tahoe, Reno, Vegas etc. TCP's flow control typically involves estimating the available bandwidth over the network (through slow-start algorithm) and retransmitting lost or corrupted data. For additional material on the features and functionality of TCP, we refer the interested reader to an excellent reference [24]. Modifications to TCP to improve its performance over wireless is an active research area [3,4,21] and [14] provides access to the activities of PILC working group of IETF which has several resources relevant to TCP over wireless networks.

3.3. Functionality of RLP protocol

The IS-95 [16] system and its evolutions, i.e. IS-95B and IS-2000, use a NAK (Negative Acknowledgment) based link layer recovery protocol in order to provide the improved error performance on the radio link. This link level protocol is called the Radio Link Protocol (RLP) [2]. The RLP stores the IP packets received in a *new data buffer* and depending on the air-link data rate, fragments the IP packets into RLP frames. It also stores a copy of every frame transmitted in a *retransmission data buffer,* in case the RLP receiver requests a particular frame to be retransmitted. The RLP receiver stores the received frames in a *resequencing buffer* and after recovering the missing RLP frames, assembles the IP packet and delivers it to the higher layers. The size of retransmission and resequencing buffers is a complicated function of a number of parameters. These parameters include the delays on the air-link and the Internet, the application being used (HTTP, FTP etc.), in case of HTTP, the number of parallel TCP connections, TCP segment size, air-link dates etc. A method for sizing these buffers is explained in [18], and in section 5.1.4 we provide the distributions of these buffer sizes.

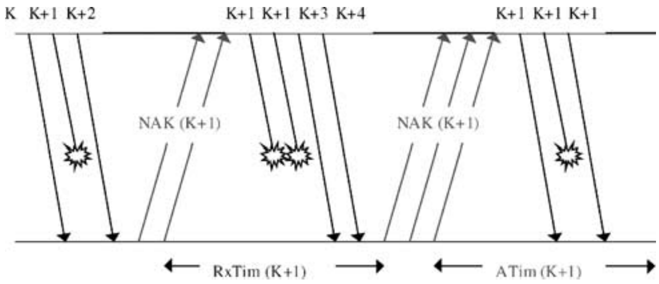


Figure 3. Graphical description of the functionality of RLP protocol.

In a NAK-based RLP, whenever a missing RLP frame is detected, the RLP receiver sends a predetermined number of NAKs specifying the missing frame number to the RLP sender. The NAKs are spaced in time to provide time diversity on the wireless channel. The RLP sender retransmits the frame with sequence number specified in the NAK control frame. In the $(2, 3)$ NAK scheme specified for RLP3, two rounds of retransmissions are performed as shown in figure 3. In the first round, the RLP receiver sends two NAKs specifying the sequence number of the missing RLP frame and activates a Retransmission Timer (RxTim). The RLP sender retransmits the frame with the sequence number specified in the NAK control frame upon receiving each NAK control frame. If the missing frame is not received and the RxTim expires, the receiver sends 3 NAKs for the missing frame and activates an Abort Timer (ATim). The sender again sends the missing frame for each of the NAKs received. If the missing frame is not received until the expiry of the abort timer, the RLP will deliver the data to the higher layer with the missing data (hole).

The timers used by the RLP are all frame counters. The timer is incremented only when a data frame is received. There is a problem with timers based on frame counting. Suppose that the sender has sent the last frame of data, if this frame is in error, the receiver sends a NAK and starts the appropriate timer. However, the sender has finished transmitting data, so the receiver may have to wait forever for the timer to expire. For this reason, RLP mandates transmission of *Idle Frames* when not transmitting data. The RLP receiver uses these frames for incrementing the timers.

3.4. Functionality of the MAC protocol

The MAC layer specification of IS-2000 is relatively more open than the RLP specification, and hence, we make spe-

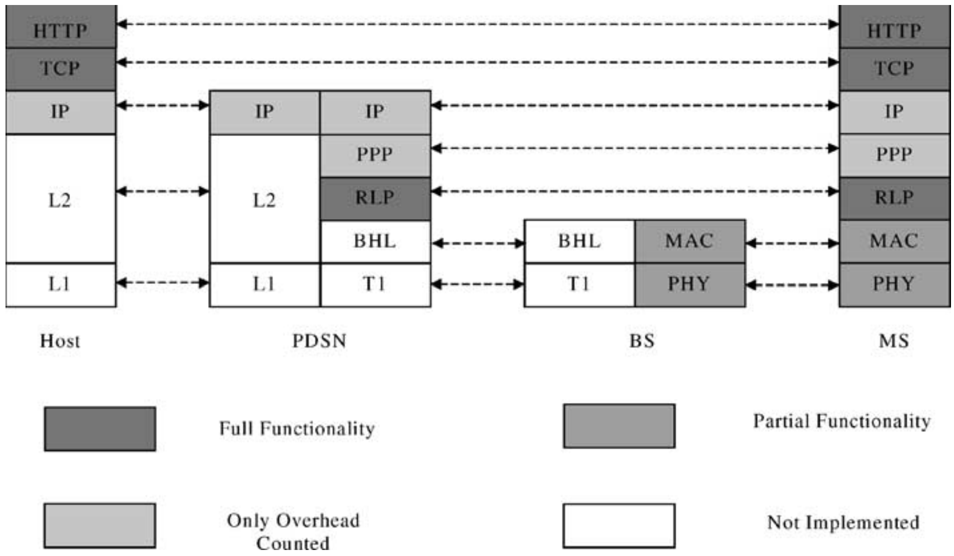


Figure 4. Protocol level description of the simulation setup.

cific assumptions on the parameters and algorithms used. The MAC protocol performs three important functions: controlling the access of data services through MAC control states, providing best effort delivery through RLP, enforcing the negotiated QoS level by appropriately prioritizing access requests. To meet the aggressive requirements of providing very high speed data services among contending users, IS-2000 MAC incorporates 2 states in addition to the TIA/EIA-95-B states of active and dormant. The two additional/intermediary states are the control-hold state and suspended state. These additional states are needed due to the excessive interference caused by idle users in the active state and to the relatively long time and the system overhead required for a transition from the dormant state to the active state. Transitions between MAC states can be indicated by MAC control signaling or by the expiration of timers. By carefully choosing the values for these timers, the IS-2000 MAC can be adapted to a wide variety of data services and operating environments. For more details on the functionality of IS-2000 MAC protocol, we refer the interested reader to [1,20].

4. Description of the simulation setup

A protocol level description of the simulation setup is provided in figure 4. In our simulation study, we consider the scenario where mobile users are accessing web pages from a fixed host. We do not simulate voice traffic. A mobile user initiates the page transfer through an HTTP request message. The serving base station receives this message and transfers it to a PCF node. The PCF node routes the message to the fixed host, through the PDSN node and the Internet. In IS-2000, the functionality of a base station (BTS) is to perform radio link related operations like power control, modulation, coding etc. The controlling functions like call admission, resource allocation, ARQ etc., are a responsibility of the BSC and PCF nodes. The transportation of user traffic between PCF and PDSN is done through the GRE/IP protocols. We assume that the connection between PCF and base station and PCF and PDSN is point-to-point. Hence, the PPP protocol runs between the mobile station and the PDSN. The PDSN node acts as an edge router in transferring mobile users' IP packets to the host. It is to be noted, however, that with proposals in the standards bodies for IP based RAN and All-IP, the point-to-point connections could be transformed to routed networks. In that case, the PPP assumption may need to be changed accordingly.

4.1. Modeling the radio access network and the Internet

The messages from and to the mobile experience delays in the radio access network (RAN) and the Internet part. The delays in the RAN typically include physical layer processing between the mobile and base station like coding/decoding delay, interleaving/de-interleaving delay etc. These delays are in addition to the data transmission delay, which is a direct function of the air-link data rates. The data from/to the base station is transported to/from the PDSN node through low bandwidth pipes. Moreover, a PDSN node serves several base stations, hence, there will be some queuing delays on this link. We model the combination of this delay and the processing delays on the physical layer as a fixed delay component, represented as delay D shown in figure 5.

The data path between PDSN and the host traverses the Internet and hence can contribute to additional variable delay. This data path is also prone to potential loss of IP segments resulting from congestion in Internet routers. We model the delay in the path between PDSN and the host as another fixed component of delay, D_{hi}, shown in figure 5. In our modeling, however, we ignore any losses that may occur in the communication path from base station to host and vice versa. This is due to the fact that, the air-link is prone to more intense loss rates than the other parts of network. We model the errors

Figure 5. Modeling the communication network between the mobile and the host.

on the air-link in detail by taking into consideration, factors such as correlation in the frame errors resulting from users' mobility [17].

4.2. Modeling the air-link between the BS and MS

The three important components implemented for the air-link are medium access control (MAC), Radio Link Protocol (RLP) and physical layer. We have implemented the radio link protocol in strong accordance with the RLP3e specification of IS-2000. The physical layer, as described above, is modeled according to the average channel quality and mobility. The mobility model assumes that users are uniformly distributed in a cell. We assume all the users to be moving at a pedestrian speed of 3 km/h experiencing the worst case fading of single path Rayleigh. The users' location is assumed to be fixed during the transfer of a web page and it changes during the user think time. Depending on the location of the user, we use data from link level simulations to estimate how much power resource the user needs for a particular data rate. Assignment of the limited air-resource (power) to the contending users is done through a sophisticated algorithm used at the MAC layer, as described in section 3.4.

Table 1
Summary of simulation parameters.

Parameter	Symbol	Value
TCP version		Reno
TCP segment size	MSS	576 Bytes
TCP max. receiver buffer size	W	32 KB
Number of parallel TCP connections	C	1
Radio link data rate	R	9.6 Kb/s to 153.6 Kb/s
RLP frame size	S_{rlp}	24, 384 bytes
Number of rounds of NAKs	R_{nak}	3
Number of NAKs in each round	N_{nak}	1, 2, 3
One-way radio link latency	D_{ll}	100 ms
One-way Internet delay	D_{hi}	80 ms
RLP NAK guard time	G	60 ms
Frame error rate	FER	1%

Table 2
Workload summary of HTTP from measurements in [22].

Workload parameter	Statistics
HTTP request size	One large peak around 250 Bytes and a small peak around 1 Kbyte.
Number of objects per web page	Mean = 4, Median = 2
File size	Mean = 6.75 KB
HTTP client think time	Mean = 30 s

5. Simulation results

In this section, we evaluate the 3G-1X system in terms of aggregate throughput and page throughput performance. We also provide the results for the user's channel ON time and RLP buffer size distributions. A summary of the simulation parameters used for this study is given in tables 1 and 2. We first give some single user performance results and then provide the results for multiple users.

5.1. Single user HTTP performance

5.1.1. Normalized object delay
Normalized object delay is a meaningful metric that provides the download time per Kbyte. This metric can be regarded as the inverse of throughput (goodput). In our simulation, object download time is obtained by noting the time elapsed from the moment a mobile generates an HTTP object request to the time it successfully receives the object. Note that this delay includes queuing delays, transmission delays, propagation delays and RLP/TCP delays for recovering corrupted data. Normalized delay is then obtained by dividing the download time by the object size.

We observe that the web object transfer delays are higher when simultaneous parallel TCP connections are allowed ($C = 4$) compared to the case of a single connection ($C = 1$). In the case of a single connection, the TCP connection for the next object in the web page is initiated only when the previous object transfer is complete. Therefore, a single connection consumes the whole available bandwidth at a time, thus reducing the web object transfer delays. On the other hand, with parallel simultaneous TCP connections, the air link bandwidth (38.4 Kb/s) is shared by more than one connection at a time. This increases the web object transfer times while reducing the web page transfer times. For example, for objects of size 1–10 Kbytes with $C = 1$ and independent errors, the average web object transfer times are 2.29, 2.76 and 3.45 s at FER of 0.01, 0.05 and 0.10, respectively. For the case of multiple parallel connections the delays at 0.01, 0.05 and 0.10 FER are respectively 3.31, 4.97 and 5.0 s.

Figure 6 shows the normalized object delay at 38.4 Kb/s and 153.6 Kb/s for i.i.d. errors. It can be noted that the normalized delay is exceptionally high for small objects (<1 Kbyte). We also observe that the delay decreases as the object size increases. This is due to the fact that for small objects, a significant component of the delay is HTTP request/response time, while for larger objects the main com-

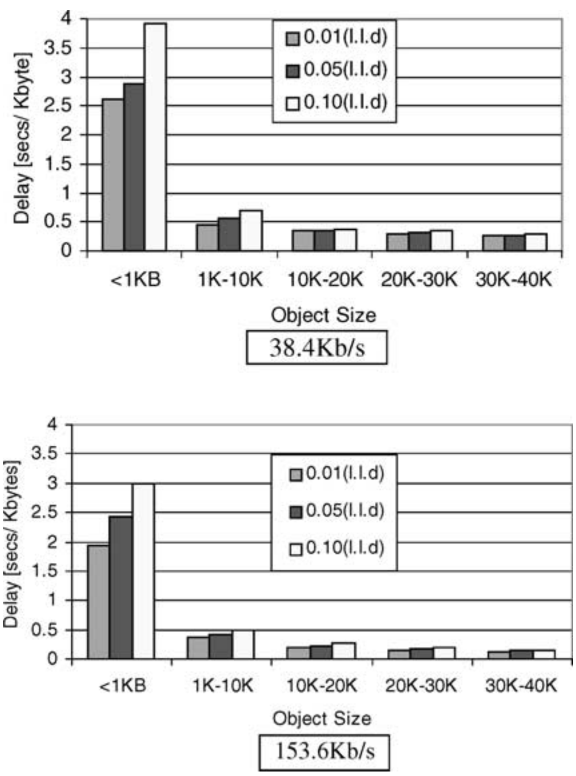


Figure 6. Web object normalized delay for a single user, i.i.d. errors, $C = 1$, 38.4 Kb/s and 153.6 Kb/s.

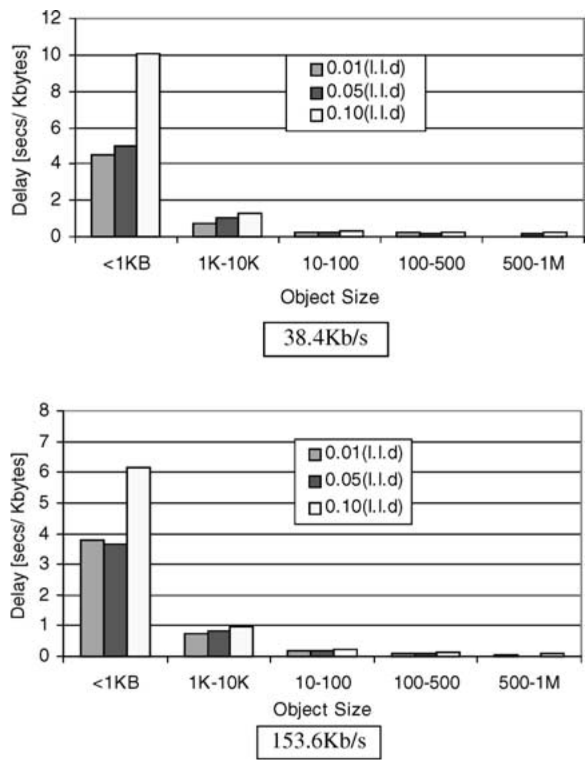


Figure 7. Normalized web page delay for a single user, i.i.d. errors, $C = 1$, 38.4 Kb/s and 153.6 Kb/s.

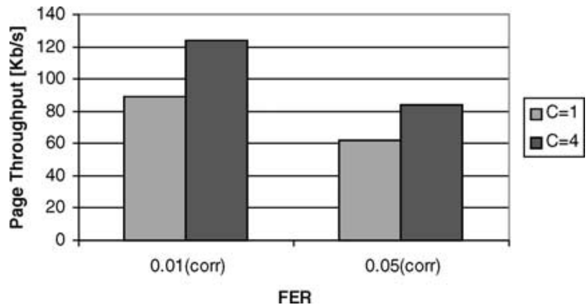


Figure 8. Page throughput for a single user and 153.6 Kb/s and different FERs and C values.

ponent of the delay is the object transmission time. Moreover, for small objects the TCP congestion window (cwnd) does not grow due to the inherent TCP slow start. At 153.6 Kb/s the normalized delays are significantly lower for larger objects compared to the case of 38.4 Kb/s air link data rate. However, for smaller objects the difference in delay for 38.4 Kb/s and 153.6 Kb/s is very small. This is due to the fact that higher radio link rates decrease the overall transmission times but the HTTP request/response time undergoes little change. The main component of delay for HTTP request transfer is one-way 100 ms link-layer delay and one-way 80 ms Internet delay.

5.1.2. Normalized web page delay

Normalized web page delay is defined similar to normalized object delay. In place of noting the download time for objects individually, we note the total download time for the page and then divide it by the page size. Figure 7 shows the normalized web page delay. The maximum number of simultaneous TCP connections is limited to one ($C = 1$). Figure 8 shows the web page throughput for very large size web pages (100–500 KB). The link data rate is fixed at 153.6 Kb/s and the results are given for single ($C = 1$) and multiple parallel connections ($C = 4$). The goal here is to see the maximum achievable throughput at a link rate of 153.6 Kb/s. For very large web pages, the achieved throughput is close to that for a long FTP application [19]. We note that the throughput is higher when up to 4 simultaneous TCP connections are allowed. As discussed earlier, multiple parallel connections favor web page throughput while individual web objects transfer delays are reduced with $C = 1$.

Note from figures 6 and 7 that the normalized delay for pages in the range of 0–1 Kbyte is higher than that for objects in the same range. This is because of the fact that every page request results in a response from the host, as described in section 3.1. The client then requests the objects/files in the page. The page size includes both the response size and the object sizes. In HTTP/1.0 model we used, the distribution of the response size is same as that for the object size. This results in making the minimum number of objects per page to be 1. Moreover, the number of objects per page is drawn from a distribution whose minimum value is 1. Hence, each page can be regarded as having a minimum of 2 objects. Since the minimum number of objects per page is 2 (which is also the median) and the mean is 4 (table 2), it is justifiable to have higher normalized delay for pages than for objects. Also,

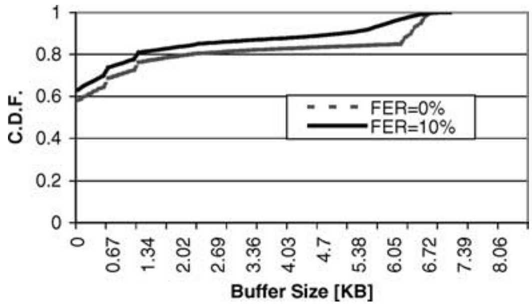


Figure 9. New data buffer size distribution for $R = 38.4$ Kb/s, MSS = 576 Bytes, $C = 1$ and $W = 8$ KB.

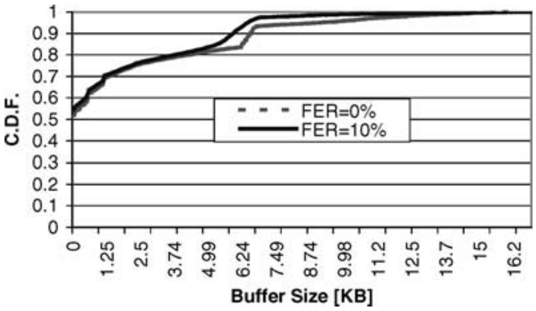


Figure 10. New data buffer size distribution for $R = 38.4$ Kb/s, MSS = 576 Bytes, $C = 4$ and $W = 8$ KB.

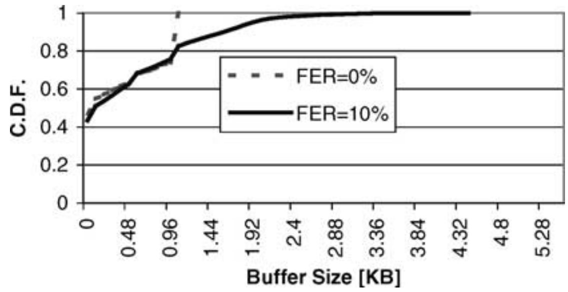


Figure 11. RTX buffer size distribution for $R = 38.4$ Kb/s, MSS = 576 Bytes, $C = 1$ and $W = 8$ KB.

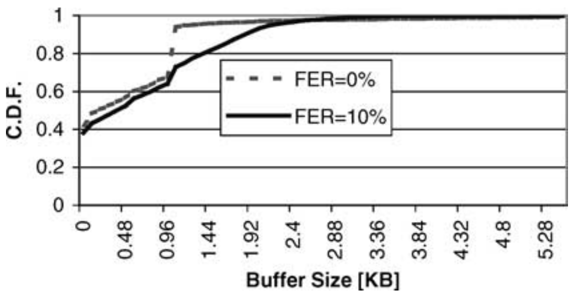


Figure 12. RTX buffer size distribution for $R = 38.4$ Kb/s, MSS = 576 Bytes, $C = 4$ and $W = 8$ KB.

note that this observation about higher normalize delay for pages than objects, does not just apply to objects and pages in 0–1 KB range.

5.1.3. Penalty in transfer delay due to frame errors

It can be noted from figures 6 and 7 that air-link errors have significant impact on object/page transfer time. Moreover, it can be seen that the penalty in the transfer time is higher for smaller objects/pages. This is due to the fact that when the objects are small, the transmission time for the TCP segments is small. When the RLP receiver sends NAKs requesting retransmissions, there is a high chance for the RLP sender to be idle and hence the retransmissions add *additional* delay of the air-link RTT. When the object sizes are larger, there is a very high chance that the retransmission requests come while there are still some new transmissions going on. So, the additional RTT on the air-link is not added. Hence, as the FER increases, there will be more additions of RTT on the air-link. Hence, although the fraction of retransmissions is the same for all object sizes (this fraction is equal to the FER), it is the non-overlapping of the retransmissions with the new transmissions that penalizes small objects more than large objects.

5.1.4. Single user RLP buffer size distribution

As described in section 3.3, RLP sender uses two buffers to perform the ARQ operation. The new data buffers store the IP packets and retransmission buffers store the RLP frames transmitted. These buffers are circular buffers and since the ARQ scheme is of selective repeat type, the circular buffer size has to be at least twice the sequence number space [6]. In this section, we provide the occupancy distributions for these two buffer sizes for HTTP/1.0 application. For more results on RLP buffer occupancy for HTTP and FTP applications for different data rates, network delays etc., please see [18].

Figures 9 and 10 depict the new data buffer size CDF for $C = 1$ and $C = 4$, respectively, at 38.4 Kb/s and a TCP window size of 8.0 KB. We note that with a single TCP connection ($C = 1$) for an HTTP page transfer, the new data buffer is empty for 58% and 63% of the time for the case of no errors and FER of 10%, respectively. This is due to the fact that with frame errors, the transmission time increases and most of the time data stays in the RTX buffers. For example, if we set the new data buffer size to 5 KB, the new data buffer will be overwritten for 16% and 10% of the cases for FER = 0% and FER = 10%, respectively. A new data buffer size of 7.0 KB will be sufficient to avoid any overwriting.

With multiple parallel TCP connections (figure 10), the required buffer size increases for the same overwriting probability. For example, fixing the buffer size at 5.0 KB will result in overwriting in 18% and 16% of the cases for FER = 0% and FER = 10%, respectively.

The CDF for retransmission buffer size with an HTTP application is given in figures 11 and 12. They show the CDF for $C = 1$ and $C = 4$, respectively, at 38.4 Kb/s and TCP window of 8 KB. For the case of no errors, the maximum retransmission buffer size is equal to an air link RTT ($2D$) worth of data. This is assuming that the state variable used to flush the retransmission buffers is sent on the reverse link every 20 ms. With $D = 100$ ms and data rate of 38.4 Kb/s, the maximum retransmission buffer size for the case of no errors turns out to be 0.96 KB. With no errors and $C = 4$, the retransmission buffer size could be greater than 0.96 KB when the state variable transmission is delayed on the reverse link because some HTTP requests needs to be transmitted.

When the errors are present on the air-link, the effective air link RTT increase due to multiple rounds of retransmissions needed for successful transmission of a data segment. The maximum retransmission buffer size in this case is given by the equation below:

$$\max B_{\mathrm{r}} = \min\bigl\{W, \bigl[(2n+2)D + nG + 1\bigr]N_{\mathrm{rlp}}S_{\mathrm{rlp}}\bigr\}.$$

In the above equation, n is number of NAK rounds, D is one-way transmission time from the RLP sender to the RLP receiver, G is the RLP NAK guard time, N_{rlp} is number of RLP frames per 20 ms and S_{rlp} is RLP frame size in bytes. For a 3-round NAK scheme used in the simulations, the maximum retransmission buffer size is limited to 4.8 KB. Note that the maximum retransmission buffer size is a function of the number of rounds of NAKs and the air link RTT. However, it is independent of the error rate. This is due to the fact that in a NAK-based RLP scheme, the RLP receiver aborts after the maximum number of retransmission attempts have been reached. The state variable is then pointed to the next missing RLP frame (hole).

5.2. Multiple user results

5.2.1. System throughput and capacity

System throughput, or the goodput, is obtained by summing the size of each web page downloaded by each user, and dividing with the total simulation time. Note that the simulation time does not just refer to the download times used in sections 5.1.1and 5.1.2 but it also includes the time when the users are in the state of reading the web page they downloaded. Since we simulate the scenario of using one IS-2000 carrier in one sector of IS-2000 cell region, the system throughput we provide in this section will indicate the Kb/s achievable per carrier per sector.

Capacity is defined as the number of users per carrier per sector. Capacity of 3G-1X system for handling data traffic is a soft metric. For a given data traffic model, protocol parameters, network delays, algorithm for resource allocation etc., the number of data users in the system determine the system attributes like average page download time. As the number of users increases, the penalty in download time increases. Defining the capacity of 3G-1X system is equivalent to putting a bound on the maximum tolerable delay which is in turn equivalent to defining the QoS metric. It has to be noted, however, that the number of maximum users allowed in the system is hard bounded by the Walsh code limitation and hence the capacity of the system is a value lower than that. Since the definition of QoS is different for different applications, service providers etc., we do not go up to saying "this is the capacity of 3G-1X system". Rather, we show how the system handles multiple data traffic sessions by providing metrics like system throughput and the average page download times as we increase the number of users.

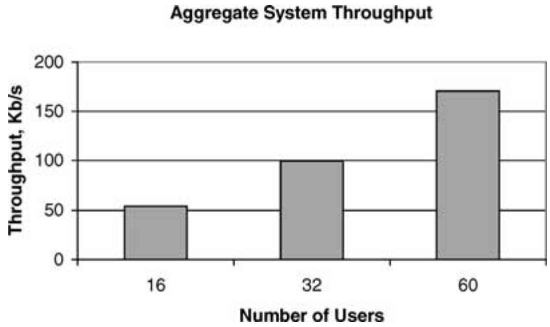


Figure 13. Aggregate system throughput as a function of number of parallel HTTP users.

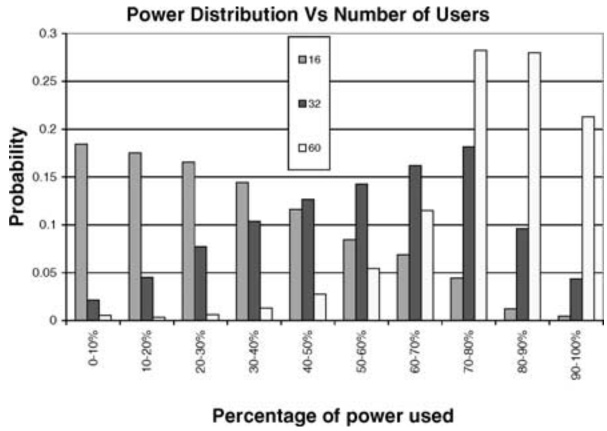


Figure 14. Distribution of the total power used by the HTTP users.

Figure 13 shows the effect of increasing the number of HTTP users on the system throughput. We see that when the number of users doubles from 16 to 32, the aggregate throughput increases by 84%, approximately doubling. This is a clear consequence of the statistical multiplexing gains resulting from the inherent discontinuous transmission in web applications, as explained in section 3.1. When the number of users is 16, the system was being under-utilized. With 32 users, the traffic has become more intense while the system was still using only 56% of the system power, as shown in figure 14. Hence, a doubling of throughput can be expected as the average traffic in the system has doubled to a value, which is smaller than the maximum traffic that can be handled. However, when the number of users increases from 32 to 60 (approximately doubling) the system throughput tends to increase by only 72%. With 60 users, the average power used is about 78%, which indicates that the system is approaching saturation. Hence, the throughput does not increase by as much as in the case of 16 to 32 users. Moreover, with 60 users, there is higher contention for the power resource and as a result the users end up with only a small fraction of the available power. With this small power, the peak data rate that a user could be assigned also goes down. This narrow bandwidth is detected by TCP's flow control and it results in TCP sending data at the lower rate.

The ultimate result of increasing the number of users in the system is a reduction in the effective data rate achieved

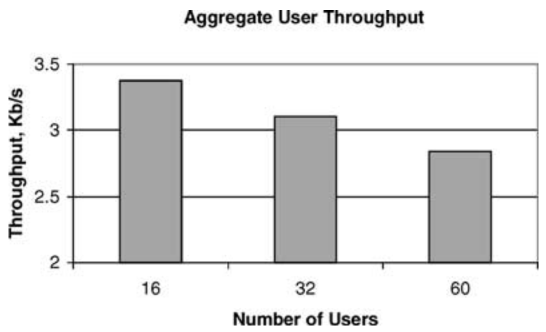


Figure 15. Aggregate throughput per user.

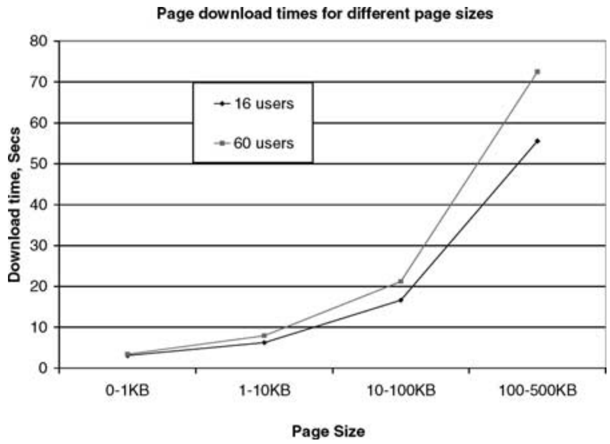


Figure 16. Web page down load times for different page sizes and parallel HTTP sessions.

by a user. This result of increasing the load in the system can be verified from figure 15, which shows the decrease in the aggregate throughput per user decreasing, as the number of users increases. Note that the aggregate user throughput is obtained by dividing the aggregate system throughput by the number of users.

5.2.2. *Average page throughput*

We now focus on the performance of individual users. We provide the individual performance by capturing the average page download time for the pages within a particular range, across all users. This type of segregation of the pages based on their sizes is required because of the heavy tailed distribution of the web objects. Moreover, such separation of pages will provide insight into the relationship between the page transfer time and the page size.

Figure 16 shows the web page download time with 16 parallel HTTP sessions running. It is important to observe from this figure that although the page sizes (plotted on the X-axis) are increasing exponentially, the page transfer times (plotted on the Y-axis) are only increasing linearly. This effect is, again, attributed to the fact that any web page transfer involves the fixed delays of request/response type of delays. In addition to these delays, the TCP flow control introduces additional breaks in the communication.

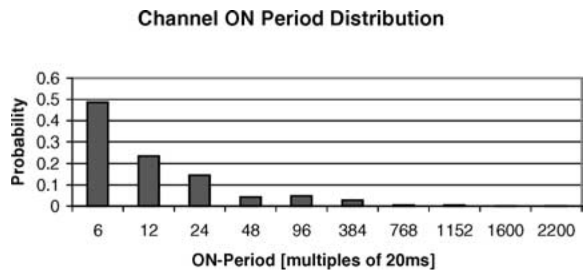


Figure 17. Distribution of channel ON time distribution with 60 parallel HTTP sessions.

For small web pages, this fixed delay component is a significant fraction of the total transfer time. However, as the objects get larger the web pages grow larger, and the TCP flow control remains in a steady state of sending the maximum-congestion-window amount of data in an RTT time. Due to this continuous supply of data from the TCP sender, the page transfer time is essentially a function of the average air-link data rate assigned by the MAC layer. With higher loading, we expect the page transfer times to increase due to increased contention for the radio resources.

This effect can be verified from figure 16 which also shows the average page download time with 60 parallel HTTP sessions. We observe that for the range of system loading considered, an increase in the number of users in the system by several factors results in a much smaller fraction of increase in page download times. For pages in 1–10 Kbyte range the download time increases from 6.24 s to 7.93 s (increase of 27%) when the number of users increases from 16 to 60 (increase of 275%). For pages in the range of 10–100 Kbyte, the download time increases from 16.6 s to 21.2 s (increase of 28%). When the system is more heavily loaded than 60 users (assuming that Walsh codes are still available), we expect the increase in delay to be more substantial than in the case above.

5.2.3. *User's channel ON time*

The channel ON time refers to the amount of time the base station is actively transmitting data to the user. This is an important system attribute which captures the dynamics of multi-user data communications. Figure 17 captures the distribution of channel ON time with 60 parallel HTTP sessions. Since the object sizes are heavy-tailed, we should expect the channel ON times to be heavy tailed also. This can be verified from figure 17. We also observe that about 50% of the ON times are about 120 ms (6 frames) long. This is again, an artifact of the TCP slow-start mechanism explained earlier. Since the mean object size is only about 7 Kbytes and the maximum TCP receiver buffer size is 32 Kbytes, most of the objects are transmitted while TCP is still in the slow start. Moreover, due to errors on the air-link, the amount of time a channel is being used by the base station for retransmissions should also be counted. Retransmissions typically need only a few frames. TCP slow start combined with air-link errors results in a significant number of transmission times about 120 ms as shown in figure 17.

6. Conclusions

In this paper we described the architecture and key features of 3G-1X Internet access system. We studied the performance of this system for HTTP application. We described the HTTP, TCP and RLP protocols and then provide a detailed description of the simulation setup. We first studied the single user performance of 3G-1X system. We find that with an air-link data rate of 38.4 Kb/s, one parallel TCP connection, object size of 1–10 Kbytes and independent errors, the average web object transfer times are 2.29, 2.76 and 3.45 s at FER of 0.01, 0.05 and 0.10, respectively. For the case of multiple parallel connections the delays at 0.01, 0.05 and 0.10 FER are respectively 3.31, 4.97 and 5.0 s. We observe that while multiple parallel TCP connections increase the object transfer delay, they decrease the page transfer delay. We also find that the penalty in the object/page transfer delay is higher for smaller objects/pages than larger objects/pages. We captured the occupancy distribution of the link layer buffers and find that the size of new data buffer depends on the max TCP receiver window. In case of retransmission buffer, we find that the size depends on the air-link data rate, round trip time and the NAK scheme used.

We then evaluated the performance of 3G-1X system for multiple HTTP transactions. We captured the aggregate system throughput as a function of the number of users in the system. We see that the aggregate system throughput increases from 54 Kb/s to 170 Kb/s when the number of users increases from 16 to 60. We show the decrease in the aggregate user throughput as the number of users increases. We show the distribution of the power used in the system as a function of number of users. We see that about 56% of the power is used on average when there are 32 simultaneous HTTP sessions and 78% of the power is used with 60 parallel sessions. We observe that for a given percentage increase in the number of users in the system, the page download time increase by a smaller percentage. For pages in 1–10 Kbyte range the download time increases from 6.24 s to 7.93 s (increase of 27%) when the number of users increases from 16 to 60 (increase of 275%). For pages in the range of 10–100 Kbyte, the download time increases from 16.6 s to 21.2 s (increase of 28%). We then show the distribution of channel ON time and conclude that about 50% of the transmission times are about 120 ms.

Terms used

(1x-EV):	3G-1X Evolution
AAA:	Authentication, Authorization and Accounting
ARQ:	Automatic Repeat Request
BSC:	Base Station Controller
BTS:	Base Transceiver System
CRC:	Cyclic Redundancy Check
EGPRS:	Enhanced General Packet Radio Service
FCH:	Fundamental Channel
FER:	Frame Error Rate
FTP:	File Transfer Protocol
GRE:	Generic Routing Encapsulation
HSDPA:	High Speed Downlink Packet Access
HTTP:	Hypertext Transfer Protocol
IETF:	Internet Engineering Task Force
IP:	Internet Protocol
IWF:	Inter-Working Function
LAC:	Link Access Control
LTU:	Logical Transmission Unit
MAC:	Medium Access Control
MSC:	Mobile Switching Center
MUX:	Multiplex Sub-layer
NAK:	Negative Acknowledgement
PCF:	Packet Control Function
PCM:	Pulse Code Modulation
PDSN:	Packet Data Serving Node
PDU:	Protocol Data Unit
PILC:	Performance Implications of Link Characteristics
PPP:	Point-to-Point Protocol
RAN:	Radio Access Network
RLP:	Radio Link Protocol
RTT:	Round Trip Time
RTX:	Retransmission
SCH:	Supplemental Channel
SDU:	Selection/Distribution Unit or Service Data Unit
TCP:	Transmission Control Protocol
UMTS:	Universal Mobile Telecommunication System

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