VOICE AND DATA PERFORMANCE OF THE cdma2000 1XEV-DV SYSTEM

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ABSTRACT

The anticipated increase in future mobile wireless packet data services has challenged the current Third Generation (3G) standardization bodies to respond with evolved 3G system specifications capable of providing increased data throughput. The cdma2000 specification has undergone a recent evolution (1XEV-DV) with the goal of improving data throughput while simultaneously providing coexisting voice services within the same radio frequency (RF) carrier [1]. This paper describes the forward and reverse link enhancements resulting in cdma2000 Revision C and presents system level performance data under mixed voice and data scenarios.

1. INTRODUCTION

The general objective regarding end-user services during the development and standardization of the current 2G networks (e.g. IS-95) was to match the services provided by PSTN and ISDN; i.e., the target was to create a mobile telephony network, which delivers pre-defined bearer services and teleservices while being both spectrum efficient and economically attractive [1]. In short, the goal of 2G was to deliver mobile low rate circuit switched voice and low rate data.

The next goal for the cellular industry was to introduce connectivity to packet data networks via cellular systems while increasing voice capacity. This was largely accomplished with the third generation (3G) evolution of IS-95 to cdma2000. From the outset of defining 3G systems, there were clear goals not only to increase the gross bit rate over the radio, but also to improve packet-switched bearer services by introducing support for various QoS classes. Furthermore, the users' ability to communicate over a packet-switched bearer service while simultaneously engaged in a voice call (or other teleservices) was deemed to be very important. Recent trends in the Mobile IP and industry trends indicate both a stronger demand for packet data services and capacity. To satisfy this anticipated increase in packet data, it is important to increase the data throughput of 3G systems while simultaneously providing the needed voice services over the same RF carrier since a significant portion of a wireless operator's revenues are currently derived from voice services. 1XEV-DV (also known as cdma2000 Revision C) accomplished this task. This feature of 1XEV-DV allows the wireless operators to utilize their spectrum more efficiently and provides a means to balance the voice and data load in their system based on their specific needs.

2. 1XEV-DV OVERVIEW

1XEV-DV is an enhancement to cdma2000's data carrying capability targeted at providing higher rates on the forward link. The 1XEV-DV system was designed to maintain backwards compatibility to all previous versions of IS-95 and cdma2000 including the existing channels and signaling structure. An equally important feature of 1XEV-DV is that it does not require new base stations, i.e. the coverage footprint is retained. The enhancements occur at the physical layer of the specification and are controlled by the upper layers. For the purposes of this paper, only the physical layer enhancements will be summarized.

2.1 Forward Link Enhancements

1XEV-DV incorporates several new features built around its time division and code division multiplexing (TDM/CDM) capability. The data bearing traffic channel is referred to as the Forward Packet Data Channel (F-PDCH or PDCH). The PDCH is shared by the packet data users and can not undergo soft handoff (SHO). Depending upon system loading the PDCH consists of 1 to 28 code-division-multiplexed quadrature Walsh subchannels, each spread by 32-ary Walsh function. It can transmit any of a set of fixed packet sizes of 408, 792, 1560, 2328, 3096, and 3864 bits. The system has variable packet durations of 1.25, 2.5, and 5 ms. The system also employs channel-sensitive scheduling via adaptive modulation and encoding with higher order modulation of QPSK, 8PSK, and 16QAM. The system makes use of a concatenation of Forward Error Correction (FEC) coding and an Automatic Repeat reQuest (ARQ) protocol known as Hybrid ARQ (HARQ). HARQ operating at the physical layer facilitates shorter roundtrip delays as compared to those associated with higher-layer retransmission schemes employed in the Radio Link Protocol (RLP). This important attribute of 1XEV-DV reduces the probability of a data session timeout (e.g. TCP/IP) as compared to RLP retransmission delays. The system has variable code division multiplexed common control channels of 1.25, 2.5, and 5 ms with a basic user packet scheduling granularity of 1.25 ms. The control channel carrying the user's MAC ID, Encoder packet size, HARQ control information, and broadcast of available Walsh codes is referred to as the Forward Packet Data Control Channel (F-PDCCH or PDCCH). The system may use up to two PDCCHs to enable data-bearing services to two different users

[1] Submitted to International Communication Conference 2003 (ICC 2003)

simultaneously. Appropriate reverse link enhancements necessary to support forward link operation of 1XEV-DV were made as well.

2.2 Reverse Link Enhancements

As many of the services in the near future are expected to be forward link intensive, the majority of the effort in designing 1XEV-DV focused on enhancing the forward link. A subsequent release will enhance the data bearing capability on the reverse link. With this in mind, only minor additions were made to the reverse link so as to support the enhanced forward link.

To support HARQ functionality, the Reverse Acknowledgment Channel (R-ACKCH) was added to provide synchronous acknowledgements to the received forward link data packet transmissions. The Reverse Channel Quality Indicator Channel (R-CQICH) is used by the mobile station to indicate to the base station the channel quality measurements of the best serving sector. The mobile station selects the best serving sector by applying a Walsh cover corresponding to the selected serving sector. In determining the 1XEV-DV design, a significant effort was undertaken to evaluate the system performance with mixed data and voice services.

3. SIMULATION METHODOLOGY

The simulation methodology used in system simulation follows the requirements of the evaluation methodology [3]. Link level simulations are performed to determine the energy requirement to achieve target Packet Error Rate (PER) for a single radio link [4]. System level simulations are then performed using the link level results as a look-up table to calculate the system performance parameters including capacity, data throughput, outage level.

3.1 Modeling of Packet Data Channels

For each time interval, the base station decides the transmission format for the scheduled users. The transmission format includes encoder packet size, slot duration and number of Walsh codes that may be used. At the receiver side, the mobile station (MS) calculates the signal to interference ratio (SIR) and then decides if the packet is erased or not.

3.1.1 Transmission format selection

The procedure of selecting F-PDCH transmission format for each data user is as follows:

1. Calculate the number of Walsh codes and power used for the voice users.
2. For each possible slot duration, determine the overhead power required for F-PDCCH and Forward Common Power Control Channel (F-CPCCH).
3. The leftover power can then be allocated to the F-PDCH. Calculate the available C/I for F-PDCH based on the C/I feedback from R-CQICH of the target MS. The C/I feedback is filtered with an IIR filter.
4. The encoder packet size and the number of slots can be determined for a target PER of 1% based on the available Walsh codes and the available C/I for the F-PDCH.
5. In case of retransmission, the same encoder packet size as in the initial transmission is kept and the above rate selection procedure is repeated.

3.1.2 PER calculation

1. The aggregate E_b / N_t for the sub-packet is calculated as

$$E_b / N_t = 10\log_{10}\left(\frac{1}{N}\left[\sum_{j=1}^{n}\frac{N_j \cdot (E_s / N_t)_j}{\alpha_{(E_s / N_t)_j}}\right]\right),$$

where

- $(E_s / N_t)_j$ is the SIR per modulation symbol for slot j
- N_j is the number of modulation symbols in slot j
- $\alpha_{(E_s / N_t)_j}$ is the de-mapping penalty for slot j if higher order modulation than QPSK is used [3]
- n is the number of slots transmitted
- N is the encoder packet size (bits)

2. A biased coin is tossed to decide the outcome (success or failure) of the transmission based on the look-up table and the E_b / N_t is calculated.

3.2 Modeling of Control Channels

The forward link common control channels such as the Pilot, Paging and Sync channels are assumed to take 20% of the total base station transmission power during the simulation time. The control channels such as F-PDCCH and F-CPCCH are dynamically modeled in the simulation.

3.3 Proportional Fair Scheduler

For each user with data in the buffer, a priority function is computed and the user with the highest priority is scheduled for transmission. The priority function for user i at time k is computed as:

$$P_i(k) = \frac{R_i(k)}{T_i(k)^{\alpha}} f_i$$

where

- $R_i(k)$ is the potential achievable data rate. It is computed using the filtered C/I feedback from R-CQICH. Note that it is computed assuming full buffer for the MS
- $T_i(k)$ is the fairness throughput
- $\alpha = 0.75$
- $f_i = 1$ for all users, i.e., the scheduler treats all users equally regardless of their traffic types.

4. SIMULATION PARAMETERS

Both the channel models and traffic models are specified in [3] and also described in the following.

4.1 Channel Models

A channel model corresponds to a specific number of paths, path delay and power profile (ITU multi-path models), and Doppler frequencies for the paths. Table 1 lists the channel models associated with the assignment probabilities used in the simulations. The channel models are randomly assigned to the various users according to the probabilities. Due to the link adaptation nature of the 1XEV-DV system, users with low speed and single finger (e.g., Channel A) tend to perform better than users with high speed and multiple fingers (e.g., Channel C).

Table 1 Channel Models

Channel Model	Multi-Path	Fingers	Speed (km/h)	Fading	Assignment Probability
A	Ped A	1	3	Jakes	0.30
B	Ped B	3	10	Jakes	0.30
C	Veh A	2	30	Jakes	0.20
D	Ped A	1	120	Jakes	0.10
E	1 path	1	0, fd = 1.5 Hz	Rician K = 10 dB	0.10

4.2 Traffic Models

The traffic models include WAP, Near Real Time streaming video (NRT), HTTP and FTP. The detail statistics of the traffic models are described in [3]. The assignment probabilities for each of the traffic models are provided in Table 2.

Table 2 Traffic Models

Traffic Model	Assignment Probability
WAP	0.5643
NRT	0.0985
HTTP	0.2443
FTP	0.0929

4.3 Other Simulation Parameters

While there are many different parameters that can be defined in the performance evaluation [3], the most notable parameters are listed in Table 3.

Table 3 Simulation Parameters

Parameters	Value	Comment
Max C/I	13 dB	
Transmit diversity	Off	
N-channel HARQ	N=4	Physical Layer HARQ
C/I feedback delay	3 slots	
Retransmission delay	3 slots	
Radio configuration	3	Voice service
Simulation time	600 seconds	

5. SIMULATION RESULTS

For Radio Configuration 3 (RC3) without transmit diversity, the voice capacity is 14 users/carrier/sector based on the requirement from [3] according to our simulations. For mixed voice and data simulations, the system is loaded with 50% of the full voice capacity, which is equivalent to 7 users/carrier/sector. The number of data users is varied in the simulations in order to get to the target outage level.

5.1 Voice service results

5.1.1 Histogram of voice data rates

Figure 1 shows the histogram of data rates for a random selected voice user. As can be seen from the figure, the percentage of the full rate, half rate, quarter rate, and eighth rate frames follow those specified in [3] (i.e., 29%, 4%, 7%, and 60% for RC3 full rate, half rate, quarter rate, and eighth rate, respectively).

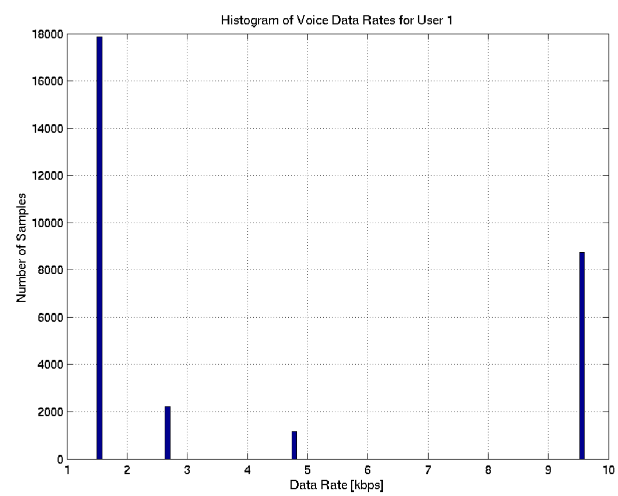

Figure 1 Histogram of Data Rates, 50% Voice Users

5.1.2 Outage probability

The outage definition for voice users can be found in [3]. The outage probabilities for all voice users are plotted in Figure 2. From the figure we observe that most of the users have outage level less than 1%. Outage typically occurs when one or more users in the same sector are in a bad radio link condition for a long period of time.

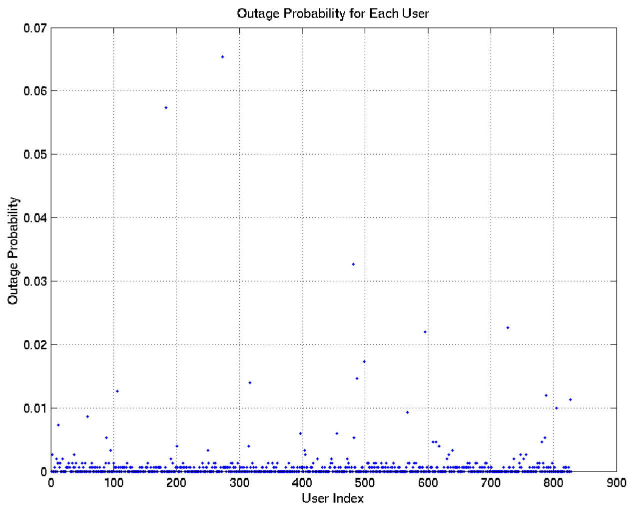

Figure 2 Outage Probability, 50% Voice Users

5.2 Data service results

5.2.1 Sector throughput and outage

The sector throughput is defined as the number of information bits that a sector can deliver correctly to all data users it serves. The outage definition for data users is specified in [3]. The data throughput and outage vs. number of data users is shown in Figure 3. As the number of data users increases, the data throughput increases from 270 to 335 kbps/carrier/sector with 50% of full voice capacity. The system outage also increases from 2.1% to 3.5%.

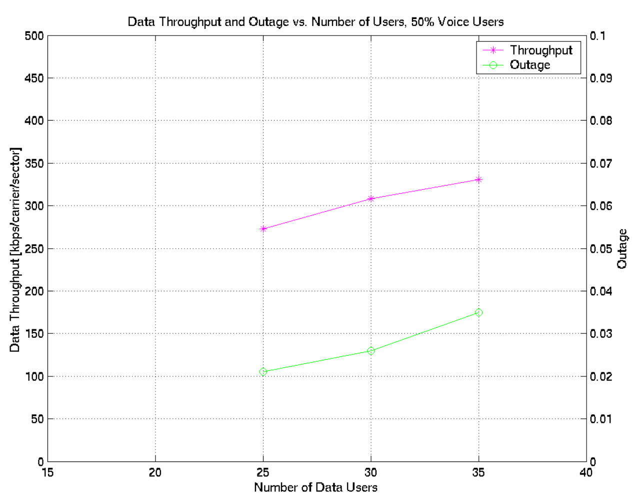

Figure 3 Data Throughput and Outage vs. Number of Data Users

5.2.2 Data throughput per user

The data throughput of a user is defined as the ratio of the number of information bits that the user successfully receives and the simulation time. The packet delay of a user is defined as the ratio of the accumulated delay for all packets of the user and the total number of packets of the user. The full picture and zoom-in picture of data throughput vs. packet delay for each user are depicted in Figure 4 and Figure 5, respectively. From the full picture we can see that FTP users normally have high data throughput as well as long packet delay. We observed from the zoom-in picture that NRT users normally have data throughput around 32 kbps with packet delay less than 200 milliseconds. NRT users with lower data throughput and high packet delay are outage users. Note that the source rate for near real time service is 32 kbps according to the traffic model specified in [3]. The data throughput for WAP users is typically less than 5 kbps with packet delay less than 300 milliseconds. The low data throughput is due to the low traffic load for WAP users.

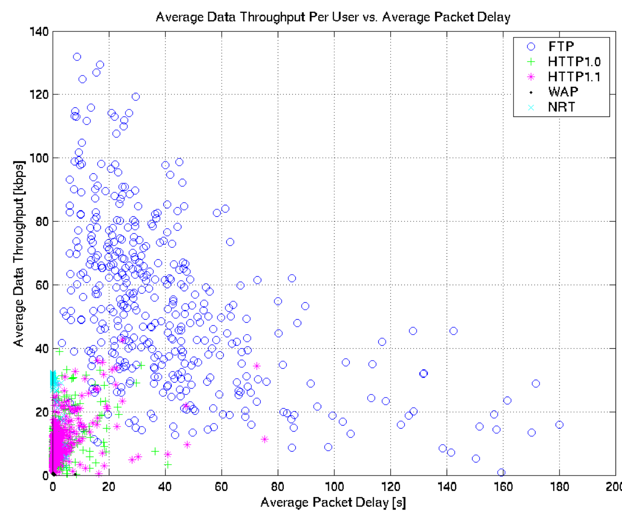

Figure 4 Data Throughput vs. Packet Delay: Full Picture

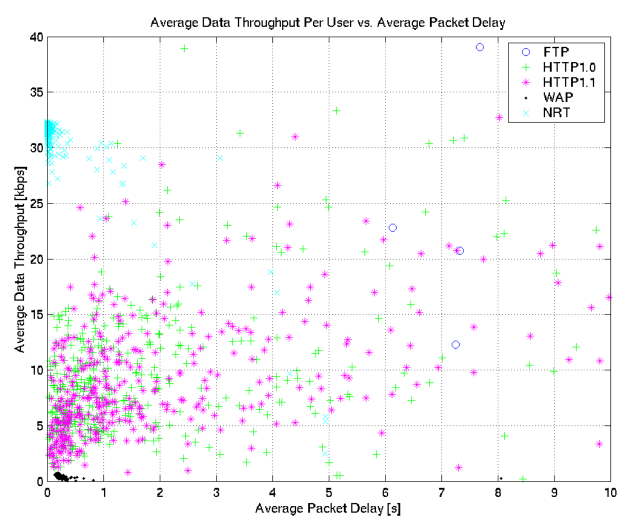

Figure 5 Data Throughput vs. Packet Delay: Zoom-in Picture

5.2.3 Packet call throughput per user

The packet call throughput of a user is defined as the ratio of the number of information bits that the user successfully receives and the accumulated delay for all packet calls of the user, where the delay for an individual packet call is defined as the time between the first packet of the packet call enters the queue for transmission and the last packet of the packet call is successfully received. The packet call delay of a user is defined as the ratio of the accumulated delay for all packet calls of the user and the total number of packet calls of the user. The full picture and zoom-in picture of the packet call throughput vs. packet call delay are illustrated in Figure 6 and Figure 7, respectively. From the full picture we can see that the packet call throughput decreases as the packet call delay increases. For FTP users the packet call throughput can be as high as 900 kbps for extreme cases. The HTTP users have relatively low packet call throughput and packet call delay compare with FTP users. This is because the packet call throughput depends on the link speed as well as the traffic load and packet arrival pattern within a packet call. The NRT users always have packet call delay of 600 seconds, this is due to the fact that for NRT users the entire simulation time is defined as a packet call. We also observed from the zoom-in picture that packet call throughput for WAP users are lower than 5 kbps with packet call delay less than 5 seconds. Again this is because WAP users have low traffic load and large packet inter-arrival time within a packet call.

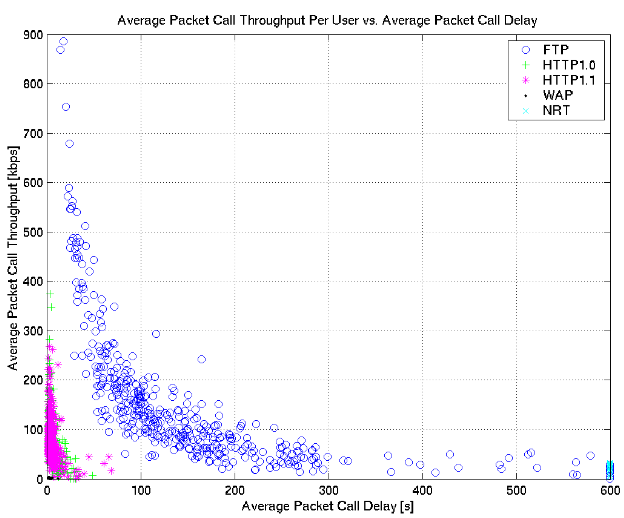

Figure 6 Packet Call Throughput vs. Packet Call Delay: Full Picture

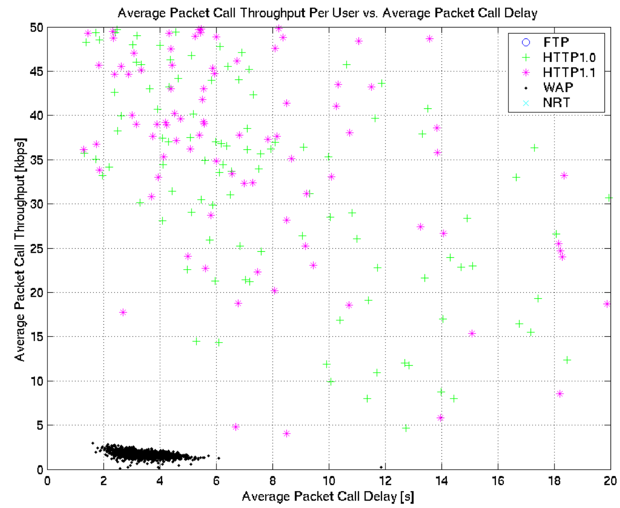

Figure 7 Packet Call Throughput vs. Packet Call Delay: Zoom-in Picture

6. SUMMARY

The 1XEV-DV system is designed to provide real-time circuit services and high data rate packet data services in the same RF carrier. The system structure of 1XEV-DV is briefly introduced in this paper. Simulation results are provided to characterize the performance of the system. The statistics of individual users with different type of traffic are also provided in this paper. We have shown that with 50% of the full voice capacity in the network, the system can still provide the sector throughput of around 270-335 kbps/carrier under the traffic models and channel models specified in [3]. The packet call throughput can be as high as 900 kbps for FTP users under extreme cases. Due to the capability of 1XEV-DV to support both voice and data service in the same RF carrier, it allows the wireless operators to utilize their spectrum more efficiently and provides a means to balance the voice and data load in their system based on their specific needs.

REFERENCES

[1] "Physical Layer Standard for cdma2000 Spread Spectrum Systems Release C", 3GPP2 Document No. C.S0002-C V1.0, http://www.3gpp2.org/Public_html/specs, May 28, 2002.

[2] "Mobile Station-Base Station Compatibility Standard for Wideband Spread Spectrum Cellular Systems", ANSI/TIA/EIA-95-B-99.

[3] "1XEV-DV Evaluation Methodology – Addendum (V6)", 3GPP2 WG5 Evaluation Ad Hoc, July 25, 2001.

[4] Yun, Youngwoo, et. al., "Reference Link Level Performance Curves of L3NQS Framework Proposal", 3GPP2 Contribution C50 20010709-025, July 9, 2001.