Performance of VoIP Services over 3GPP WCDMA Networks

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Abstract—**We analyze the performance of voice services over 3GPP WCDMA Networks for Release 99 and Release 7 deployments. We show, by using simulations, that the system capacity of Release 7 VoIP is significantly higher than the Release 99 circuit-switched voice under similar system conditions and voice quality. These results are obtained under the 3GPP simulation assumptions where uplink and downlink constraints for power, code and interference are considered. Both AMR 12.2 and 5.9 as well as different types of mobile receivers (Rake and equalizer, single and dual receive antennas) are considered. We also show that the capacity and performance can be increased significantly further by using uplink interference cancellation at the base station. The performance of best effort traffic in the presence of voice users is also analyzed.**

Keywords-VoIP; 3G; AMR; HSPA; HSPA+

I. INTRODUCTION

Voice services over wireless networks have been traditionally provided by circuit-switched (CS) systems. As more advanced 3G systems are being widely deployed, using VoIP instead becomes an alternative possibility for operators. These 3G systems allow integration of data services with voice services and thus provide higher network bandwidth efficiency, better manageability and cost savings as well as richer services.

HSDPA and HSUPA (called HSPA together) have been standardized in 3GPP Release 5 and 6 respectively. They provide significant improvement over Release 99, including data rate increase and packet delay reduction by using features such as Hybrid ARQ, Adaptive Modulation and Coding, shorter frame sizes, opportunistic scheduling and other QoS enablers. Release 7 (HSPA+) introduced further improvements including overhead channel reduction and battery savings features [1]. All of these advancements make these systems well suited not only for data applications but also for delay sensitive applications such as VoIP considered here. Although it is possible to have VoIP over even Release 99 [2], significant capacity gains over circuit-switched systems can only be achieved with HSPA and HSPA+.

Although the HSPA radio-access networks can accommodate VoIP, an end-to-end all-IP VoIP implementation requires changes at the core network as well. These include call control signaling (IMS), transport networks and handover mechanisms from VoIP to CS voice under mobility conditions. Recently in order to realize the capacity benefits of HSPA+ air interface while using the current core networks, CSoHS (Circuit-Switched Voice Services over HSPA) was introduced in 3GPP [3]. This feature achieves higher capacity of HSPA+ systems while still keeping the circuit-switched core network. The capacity of VoIP systems shown in this paper is comparable to the capacity of CSoHS under most deployment and use scenarios. In the rest of the paper, we will use the term "voice" as a generic term for circuit-switched voice, VoIP or CSoHS.

In this paper, we discuss and show that high-quality VoIP service can be provided in HSPA and HSPA+. We analyze how the system parameters and features, including interference cancellation and mobile receiver types, affect the system capacity and coverage. We show that the VoIP capacity of a Release 7 network is much higher than that of Release 99 circuit switched network under similar conditions. We include for comparison the Release 6 VoIP performance whose capacity is in between Release 99 and Release 7. The remaining data capacity of Release 7 networks is shown to be considerably higher when same numbers of voice users are served by VoIP instead of circuit-switched voice.

II. SIMULATION MODEL

We consider a cellular wireless system where mobile terminals are served by base stations (Node B). The performance and capacity of this system is investigated via simulation. The simulation framework is based on 3GPP simulation assumptions [4]. In particular, we consider a 57-cell deployment with wraparound and inter-site distance of 1 km. The chancel model is a mix of Pedestrian A 3 km/h, Pedestrian B 3 km/h, Vehicular A 30 km/h and Vehicular A 120 km/h with respective ratios of 30%, 30%, 20% and 20%.

The voice sources used in our simulations are based on Markov models of the AMR 12.2 and 5.9 vocoders [5]. A full rate voice frame is generated every 20ms when the user

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is active (talking) and a silence insertion descriptor (SID) frame is generated every 160ms when the user is not talking. The size of a full rate VoIP frame is 244 bits for AMR 12.2 and 118 for AMR 5.9. The SID frame size is 39 bits. The activity factor (the probability of being in active state) is 0.5 and the active and non-active times are exponentially distributed with a mean value of 2 seconds.

The main factors that determine the voice system capacity are the voice quality, uplink interference (system stability) and code and power limitations on the downlink. The voice quality is quantified here by the radio link delay and frame loss rate. A delay bound of 100ms is used on both the downlink and uplink such that the frames whose delays exceed this bound are discarded. It is assumed that a voice user is in outage if the total frame loss due to transmission errors and the delay requirement is more than 3%. The system capacity is reached when the number of such users is more than 5% of the total number of users. The system stability on the uplink is also determined by the total interference. We impose that the "Noise Rise" defined as the ratio of total received power to the thermal noise should not exceed 7 dB for more than 1% of the time. The system capacity on the uplink is taken as the minimum of this "Noise Rise" based capacity and the user outage capacity as defined above. For the downlink, the system capacity is determined only by the user outage which could also be caused by code limitation as will be discussed below.

For the downlink simulation, we also incorporate a receiver implementation margin. For a Rake receiver, 0.8dB is assumed to account for the loss due to RF front-end, channel estimation and other losses. For an equalizer receiver, SINR loss is added as a function of different channel types to the ideal equalizer output to accurately model the practical equalizer implementation.

A. Release 99 Assumptions

In Release 99, at the physical layer, an AMR frame is carried over the Dedicated Physical Data Channel (DPDCH). On the downlink, the DPDCH is time multiplexed with Dedicated Physical Control Channel (DPCCH) which carries the transport format combination indicator (TFCI), transmit power control (TPC) bits, and the Pilot. The power offsets of TPC, TFCI, and Pilot relative to the DPDCH are specified by PO1, PO2, and PO3, respectively. Since the DPCCH is constantly transmitted, it could consume considerable amount of power. In our simulations, we use 3dB for PO1, PO2 and PO3 for AMR 12.2. On the uplink, the DPDCH is code-multiplexed or I/Q multiplexed with DPCCH using different power setting indicated by β_d and β_c. These power settings vary according to TFC in order to maintain the same uplink SIR target.

On both the uplink and downlink, a closed-loop power control algorithm adjusts the transmission power to achieve 1% packet error rate. However, during soft-handover (SHO), the same power command (TPC) is sent to different Node Bs in the Active Set which might experience different radio conditions. Therefore the transmit powers of Node Bs may drift apart if Node B solely relies on the TPC to adjust its power. It is crucial to take into account the impact of power drift on capacity. For mitigating the power drift problem in our simulations, we consider a power balancing algorithm where RNC adjusts the transmit power of each Node B in the Active Set every 20ms based on the Node B transmitted power reported 20ms ago.

B. Release 6 and 7 Assumptions

The header overhead in a VoIP scenario is significant if no header compression is used. We assume that robust header compression (RoHC) [6] is employed which reduces the total RTP/UDP/IP header overhead to 3 bytes. The corresponding transmitted full VoIP packet size is 307 [317] bits and the SID packet size is 120 [137] bits on uplink [downlink] excluding the 24 bit CRC at the physical layer.

It is assumed that the pilot channel on the uplink (DPCCH) uses the so called "slot 1" format where 8 pilot symbols are transmitted every 2ms.

The power gains of traffic and overhead channels have an impact on the spectral efficiency on the uplink. By simulation and analysis, we have determined the optimal settings which minimize the required Eb/No. These are tabulated in Table I as the ratio (offset) of a channel's power to the DPCCH power.

On the uplink, the number of HARQ retransmissions provides a trade-off between time diversity gain and delay. We used a maximum of 5 attempts such that the power control keeps the physical layer packet error rate at 1% after 5 attempts. On the downlink, the code usage of extra HARQ attempts cancels the benefit of time diversity. Therefore the downlink packet error rate is adjusted so that 90% of packets are decoded successfully on the first attempt and a maximum of 4 attempts is used.

Release 7 introduced CPC (Continuous Packet Connectivity) which allows the uplink and downlink transmissions to take place at periodic intervals [7]. This feature reduces the transmitted power (and thus increases the UE battery life) because the UE does not have to monitor and transmit overhead channels every TTI (Time Transmission Interval). This reduction in the transmitted power also helps to increase the uplink capacity by decreasing the total interference. This improvement is especially significant when there are users who transmit data infrequently. For a VoIP user, a packet is generated every 20ms during talk and every 160ms during silence periods. By using the Discontinuous Transmission (DTX) feature defined in CPC, a VoIP UE now transmits the pilot channel only at the DTX cycle times and data packet transmissions.

A Channel Quality Indicator (CQI) value is reported by the User Equipment (UE) for downlink transmission selection and scheduling. We have observed by simulation that reporting CQI only every 16ms results in negligible reduction in the downlink VoIP capacity. For the HSDPCCH, we used a higher power offset for a user in soft-handover as shown in Table I since HSDPCHH is decoded by the serving Node B only.

On the downlink, the MAC-hs scheduler at the Node B needs to be both delay sensitive and channel sensitive. Thus it takes into account the delay constraints while still exploiting the multi-user diversity. For the results in the paper, we use a scheduler that prioritizes and allocates resources to VoIP users according to the following priority metric:

$$priority(u) = \log_{10}\left(\frac{R^{u}_{requested}}{R_0}\right) + \min\left(\left(\frac{D^{u}_{HOQ}}{d_{bound}}\right)^2, (\delta)^2\right) \quad (1)$$

The first term of the metric in (1) is the channel sensitive part that represents the channel quality estimate for a user by its requested rate obtained from the CQI information. The second term is the delay sensitive part represented by the ratio of head-of-queue packet delay of the user and the delay bound. The scaling constants are used to adjust the tradeoff provided by the two non-linear functions.

The scheduler may serve multiple users in the same TTI, bundling multiple vocoder packets for a user in a single MAC-hs packet whenever necessary. The scheduling algorithm is an empirically developed variant of a number of QOS aware channel sensitive scheduling algorithms discussed in literature [8].

C. *Advanced Node B Receiver*

We have analyzed the effect of Interference Cancellation (IC) at Node B on VoIP performance. Our IC implementation is based on [9]. We assume that a certain fraction of a UE's uplink channels can be effectively cancelled at a Node B if the UE's active set includes a cell of this Node B. The efficiency of the cancellation is determined by a factor called "beta". The "beta" is equal to the ratio of cancelled energy (subtracted from the total interference) to the total received energy. It is an increasing function of measured average Ec/Nt per antenna at Node B as shown in Fig. 5 for both 2ms and 10ms TTI. The total effect translates to a reduced total interference and noise seen by each user. The overhead channels, namely the pilot, HSDPCCH and EDPCCH are cancelled according their respective "beta" values. The cancellation of the data traffic happens only when the data packet is decoded successfully. The UEs are first sorted according to a priority metric and a decode is attempted for each UE. The IC algorithm uses two attempts to cancel a user's channels i.e. after all the UEs have been visited, a second iteration of this operation is performed.

III. SIMULATION RESULTS

We obtained the uplink and downlink system capacities by running simulations for each link where simultaneously active voice users are present in the system. An equal number of UEs are dropped in each cell randomly with uniform distribution and interferences from all UEs in 57-cells according to the wrap-around model are considered. The number of UEs is varied and the performance metrics are computed to determine capacity according to the defined criteria.

A. *Release 99 Voice Capacity*

Table II summarizes the Release 99 voice capacity. For the downlink, AMR 12.2 uses DPCH slot format 8 with spreading factor 128 while AMR 5.9 uses DPCH slot format 4 with spreading factor 256. As shown in Table II, when single antenna is used at the UE, the voice capacity is downlink limited. Using an equalizer at the receiver could improve the capacity by about 20% compared with the Rake receiver.

A straightforward way to improve the Release 99 voice capacity is to use AMR 5.9 instead of AMR 12.2. However, as shown in Table II, AMR 5.9 only provides 30% capacity gain over AMR 12.2, much less than the simple ratio 12.2/5.9 of the voice data rates. This is mainly due to two reasons. 1) Puncturing loss: After rate matching, AMR 12.2 effectively uses the rate 1/3 Convolutional code for class A bits while AMR 5.9 only uses an effective coding rate of around 1/2. The puncturing (coding) loss alone could account for 1dB extra power. 2) Power of overhead (DPCCH) channel: This is not reduced when going from 12.2 to 5.9. In order to maintain the same quality of TPC and pilot bits, AMR 5.9 and AMR 12.2 needs to use the same amount of power for DPCCH. In fact, DPCCH overhead accounts for almost half of the power that is required to support an AMR12.2 call. This computation is based on VAF of 50% and power offset value of 3dB for the pilot. Considering the above two factors, reducing the vocoder rate from 12.2 kbps to 5.9 kbps results in much less reduction in the total consumed DPCH (DPDCH + DPCCH) power by the UE.

B. *Release 6 and 7 VoIP Capacity*

As discussed previously, the main difference between Release 6 and 7 on the uplink in our simulations is the CPC feature. Since uplink is the bottleneck for the system capacity, we have chosen the CPC parameters to maximize the system capacity while still providing the required VoIP performance. The DTX cycle can be adjusted according to the voice activity of the user as defined in the standard. We assumed that the cycle length is 8 TTI for 2ms TTI and 2 TTI for 10ms TTI in both the active and non-active states. The standard also defined preamble and postamble which are pilot transmission before and after the DTX cycle times. These are used to help the channel detection and estimation at Node B. We assumed that the preamble and postamble are 2 and 1 slots respectively. Thus for the 2ms TTI, the user transmits pilot and data, if available, every 8 TTI, starting 2 slots before the every 8th TTI and finishing after 1 slot. We use a DTX cancellation mechanism where the user transmits data without waiting for the next DTX cycle if the packet delay exceeds a certain threshold. This threshold is 30ms for 2ms TTI and 0ms for 10ms TTI. We note that it is possible to use shorter DTX cycle lengths (such as 4 TTI in 2ms TTI) and not use the above DTX cancellation but our simulations and analysis showed that using a larger cycle length of 8 TTI with this DTX cancellation policy provides higher capacity gains.

Fig. 1 and Fig. 2 show the 99th percentile of "Noise Rise" for different number of users. This corresponds to the point where the "Noise Rise" exceeds the level on the y-axis for 1% of the time and the system capacity is defined when it is equal to 7 dB. The "Noise Rise" for the IC cases is measured after the cancellation takes place. In other words, it is not measured at the antenna input but it is the "Effective Noise Rise" seen by the traffic packets. The system capacity according to the user outage definition was higher than the "Noise Rise" capacity shown here and therefore are not shown separately. The capacity numbers determined from these graphs are shown in Table III.

The results show that the VoIP capacity is higher for a TTI of 2ms than 10ms. The difference is more pronounced for Release 7 where most of the gain of the 2ms over 10ms comes from using the DTX feature in Release 7. For the 10ms TTI scenario, the DTX happens very rarely when the user is talking since most TTIs are occupied by voice transmission. In contrast, for the 2ms case, the DTX provides gain in terms of reduced pilot power even when the user is talking. In this case, a packet transmission takes an average of 3 HARQ attempts (6ms) but a VoIP packet arrives every 20ms so the UE can enter DTX mode even during the talk time.

On the downlink, the VoIP system performance is significantly affected by the MAC-hs scheduler located at the Node B. In addition to the variable delay introduced by Hybrid ARQ retransmissions, the scheduler introduces queuing delays for the downlink traffic.

The vocoder packets that cannot be played out in time are effectively lost even if they are transmitted and decoded correctly over the air. Hence, for supporting high VoIP traffic the scheduler must trade-off channel sensitivity and delay sensitivity.

In section II, the outage criterion for VoIP calls and the notion of VoIP capacity was discussed. Fig. 3 shows the fraction of users in outage under that criterion, as a function of number of VoIP users per cell under three different deployment scenarios identifying significant factors which affect the downlink VoIP capacity. Table IV summarizes the VoIP capacity numbers thus obtained. The significant factors are, (i) use of Rx-diversity, (ii) use of F-DPCH channel, (iii) use of equalizers/advanced receivers.

In Release 5 HSDPA, each UE using the data channel, HSDSCH is deployed with a dedicated SF-256 channel, namely the associated DPCH (A-DPCH) for power control. As the number of VoIP users increase, deployment with A-DPCH quickly runs out of codes for HSDSCH. Therefore the VoIP capacity, although higher than Release 99 voice capacity, gets limited and is not improved much by use of advanced receivers such as equalizers at the UE.

Release 6 and 7 alleviate the code consumption problem by using the Fractional-DPCH channel. Results for Release 7 downlink capacity are obtained by using the Release 7 flavor of F-DPCH, namely Enhanced FDPCH. Release 7 VoIP capacity numbers also include the additional enhancement of HSSCCH-less operation. Other Release 7 downlink enhancements include 64 QAM modulation and HSSCCH-less operation for VoIP traffic [7].

As it can be seen from Table IV and Fig. 3, significant gains in downlink VoIP capacity are obtained by the use of equalizers in the mobile receivers instead of RAKE receivers.

The packet delay is an important metric of voice quality. Fig. 4 shows the combined uplink and downlink over the air delay at the system capacity values. These were obtained simply by adding the uplink and downlink delays. The 98% of the total delay is plotted to capture the effect of the de-jitter buffer. The graph shows that the total radio-link delay will satisfy the ITU recommendations [10] for a satisfactory voice call.

C. Mixed Data and Voice Traffic

In this section, we look at the performance of the system when both voice users and best effort (BE) users are present. The BE traffic is generated by a full buffer traffic source. It is important to quantify the amount of BE throughput when there are voice users present in the system for evaluating the overall performance of the system. In particular, we look at the downlink throughput since the bottleneck for the voice capacity is the uplink (assuming mobile receive diversity).

On the downlink, the Node B HSDPA scheduler used in the simulation is delay sensitive for VoIP traffic and gives priority to delayed VoIP users over BE users. Among VoIP users, the priority metric is a function of instantaneous channel quality and head of queue packet delay of the user. Among the BE users, the priority is decided by a proportional-fair metric. As the number of VoIP users increases, the throughput served to BE users decreases because VoIP traffic consumes both power and orthogonal code space. Additionally, if the number of VoIP users is large, significant code-space is consumed by Associated DPCH or Fractional DPCH.

The downlink application layer throughput of BE users in the presence of VoIP users are shown in Fig. 6. It can be seen that, at the VoIP capacity limited by the uplink, the downlink application throughput under Release 7 deployment and Type III receivers (Equalizer with Receive Diversity) [11] is around 860 Kb/s. The other curves in Fig. 6 show that the variation of downlink throughput of BE users under Release 5 (with ADPCH) deployments as well as Type I (Rake with Receive Diversity) and Type III receivers.

IV. CONCLUSION

3GPP Release 7 provides efficient support for both best-effort and VoIP traffic. It can support significantly higher number of VoIP users at similar QoS levels of the circuit-switched voice systems. When Interference Cancellation is employed, the capacity is three times the Release 99 voice capacity. Simulations also show that a significant amount of BE traffic can still be served on the downlink at the VoIP capacity operating point.

On the uplink, the capacity improvements of VoIP are mainly due to time diversity gain by the use of HARQ and the DTX features which reduce the total transmitted power and interference.

On the downlink, the high VoIP capacities obtained in Release 7 can be attributed mainly to the use of delay and channel sensitive MAC-hs scheduling and resource allocation, the use of F-DPCH channel, and the presence of receive diversity and advanced receivers/equalizers at the mobile.

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TABLE I . CHANNEL POWER OFFSETS ON HSUPA

Power Offset [dB]	EDPDCH 120 bits	EDPDCH 307 bits	HSDPCCH	EDPCCH
TTI = 2ms	7.11	9.92	0 (SHO), -2 (non SHO)	0
TTI = 10ms	2.92	6.02	0 (SHO) -2 (non SHO)	-5.46

TABLE II. RELEASE 99 VOICE CAPACITY RESULTS

	Downlink		Uplink
	1Rx Rake	1Rx Equalizer	
AMR 12.2	51	63	69
AMR 5.9	67	82	84

TABLE III. UPLINK VoIP CAPACITY RESULTS

	Rel. 6	Rel. 6 w/ IC	Rel. 7	Rel. 7 w/ IC
TTI=2ms	103	136	136	190
TTI=10ms	95	126	106	142

TABLE IV. DOWNLINK VoIP CAPACITY RESULTS

Deployment	A-DPCH, RAKE, Equalizer	Rel 7 F-DPCH RAKE	Rel 7 F-DPCH Equalizer
1 Rx		61	83
2 Rx	101	172	232

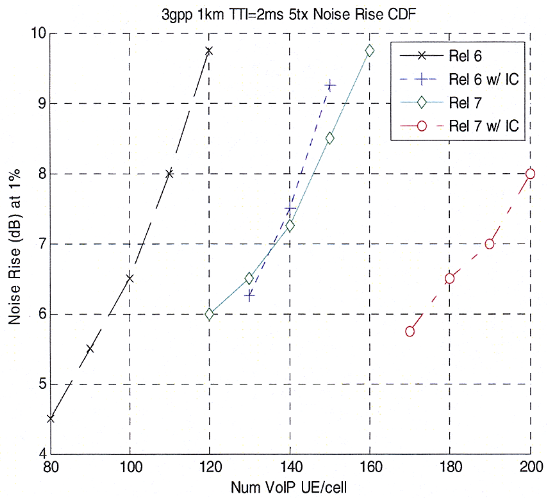

Figure 1. "Noise Rise" outage for TTI=2ms

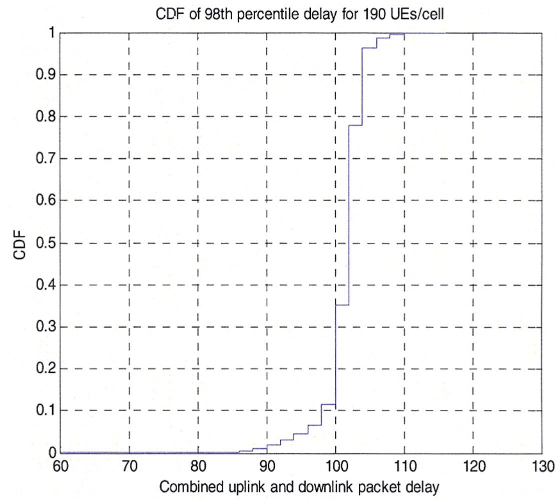

Figure 4. Uplink and Downlink Total Packet Delay

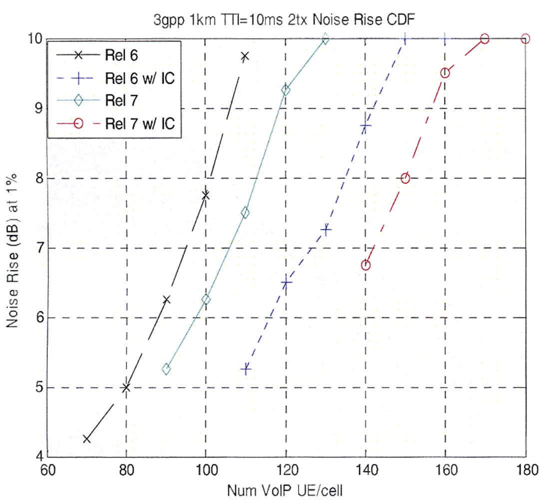

Figure 2. "Noise Rise" outage for TTI=10ms

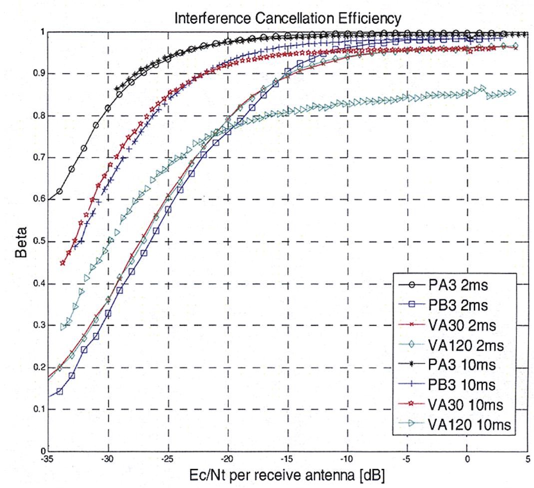

Figure 5. Interference Cancellation Efficiency

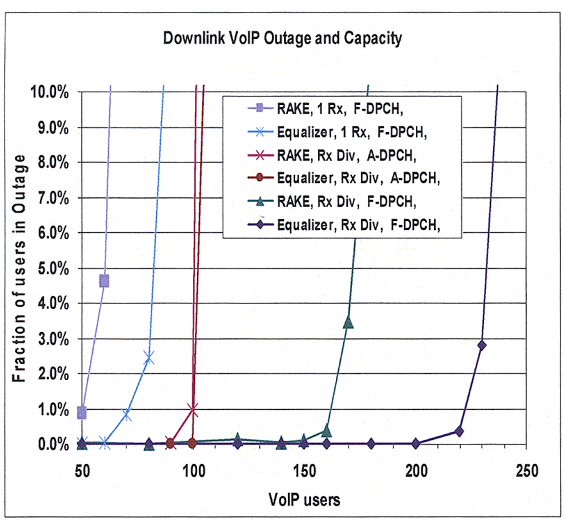

Figure 3. Release 6 and 7 Downlink Capacity

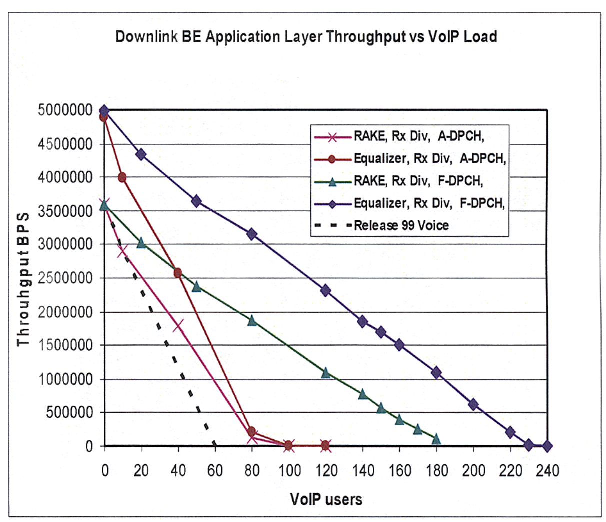

Figure 6. Downlink Mixed Voice and BE Performance