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1. Introduction

Previous contributions [3][4] have introduced voice system capacity results with DARP and Mobile Station Receive Diversity (MSRD). This contribution addresses the performance of mixed voice and HTTP traffic with EGPRS MCS-1, MCS-2, and MCS-3.

2. Simulation Details

Voice and HTTP traffic were generated in a static simulation in accordance with the system configuration description in Appendix Table 2 and the HTTP traffic model in Table 3. The HTTP traffic model is similar to that in [1], with minor changes. A dedicated 32 kbps backhaul resource was assumed for each user, and as a consequence, the network delay for each packet is a deterministic function of the packet size. Also, out-of-range (OOR) draws of random variables used in the generation of the HTTP traffic were either limited or re-cast to better match the mean values reported in [1].

For two antennas, it was shown in [3] that maximal ratio combining (MRC) and Single Antenna Interference Cancellation (SAIC) link mappings could be used to conservatively estimate the performance of Dual Antenna Interference Cancellation (DAIC). In this method, the CIR and DIR are calculated for the max-ratio sum of the outputs of the two antennas, and are then used to estimate BEP and FEP through the stage-1 and stage-2 maps derived from the non-diversity SAIC link simulations.

In the simulation, 3 time slots were reserved for data. Handsets were limited to a single receive slot for simplicity. In the absence of this restriction, we would expect the user throughput to increase with the number of slots, and the relative performance gains of DARP and MSRD to remain unchanged. All sites were assumed to be time synchronized. However, because voice and data slots may have significantly different loadings, the time slots reserved for data at each site were chosen randomly to provide a common interference environment for the voice and data slots.

The simulation used MCS-1, MCS-2, and MCS-3, but without Incremental Redundancy (IR). IR may be added to the simulation in the future. Link Adaptation was based on a filtered measure of FER, to avoid speculative decoding of multiple MCS rates.

A mix of 70% voice and 30% HTTP was used, where the percentage denotes the fraction of total population using the particular application. For circuit voice traffic alone, the (Voice) Effective Frequency Load (EFL) was defined as the number of current voice users divided by the total slots (frequencies x slots) in a sector. It may be useful to consider the circuit voice load to be "reservation Erlangs", and define an associated "interference Erlangs" as the reservation Erlangs reduced by the voice activity factor. Similarly, an effective interference load can be associated with HTTP and FTP calls, though the relationship is not fixed because the total number of times slots associated with a call depends on the (M)CS. Thus, for any loading of voice and data traffic, we define the Effective Interference Load as the average fraction of slots which are occupied by either voice or data.

3. System Performance

In this contribution, the system voice capacity is defined as the Effective Frequency Load (EFL) at which 98% of the calls have less than 2% FER over the call duration. Blocked calls are counted against the call satisfaction statistics. The performance metric for HTTP is the average of the per-user throughput. The reading time for packet calls is not included in the calculation of throughput.

Figure 1 shows the average per-user data throughput of a mixed voice-data system, in flat fading at 50 km/hr. The concurrent voice capacity of the system in the presence of the data traffic is illustrated in Figure 2.

Figure 3 and Figure 4 are similar to Figures 1 and 2, but differ in that user throughput and voice satisfaction are shown against the Effective Interference Load instead of the (Voice) Effective Frequency Load (EFL). The "DARP – Voice Only" curves in Figures 2 and 4 denote the result of previous system simulations without HTTP traffic. By referring to Figure 4, it is apparent that the impact of interference on voice performance is represented better by using the Effective Interference Load.

In the Figures, results are presented for the conventional receiver (CR), 1-antenna DARP, and DARP+MSRD. The DARP+MSRD receiver is shown with the combined antenna impairments of 2 dB antenna gain imbalance (AGI) and an antenna correlation of 0.4, where the antenna correlation is defined here as the magnitude of the complex correlation. Note that the curves with DARP+MSRD use the conservative MRC+SAIC approximation for the performance of dual antenna interference cancellation, as presented in [3].

Table 1 contains a summary of the Figure 1 and Figure 2 curves. The voice capacity is defined as the EFL at which 98% of the calls have an FER <2%. For data, the average per-user throughput (for 1 slot) is compared at an EFL loading of 20%.

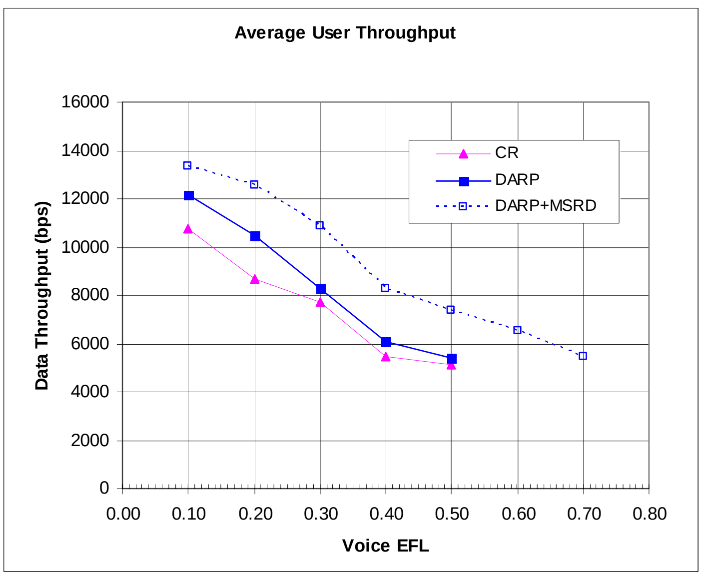

Figure 1 – User data throughput versus Voice EFL, 50 km/hr

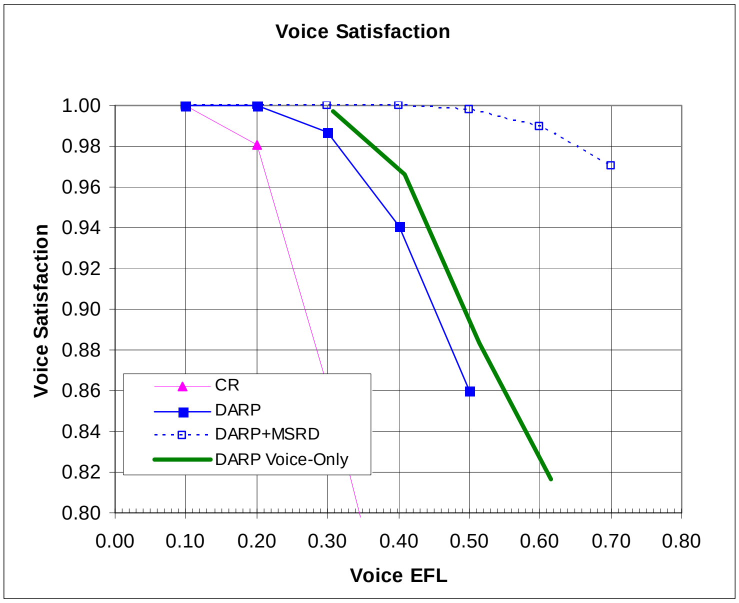

Figure 2 – Voice satisfaction versus Voice EFL, 50 km/hr

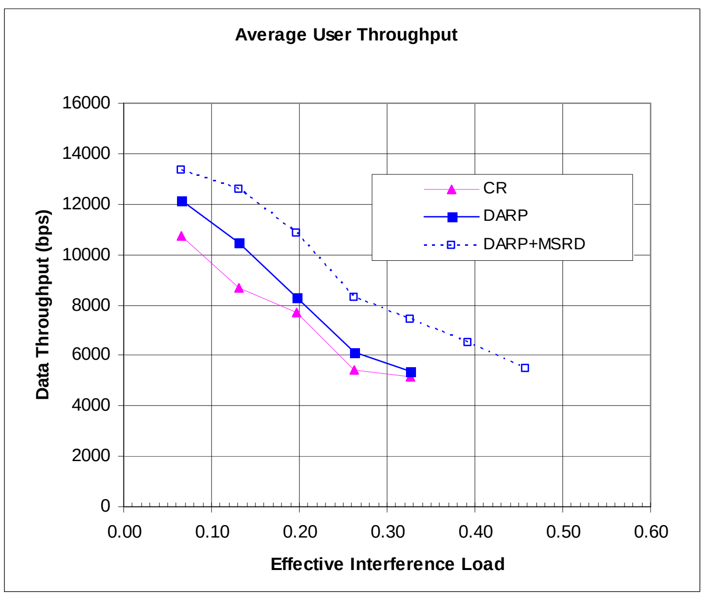

Figure 3 – User data throughput versus Effective Interference Load, 50 km/hr

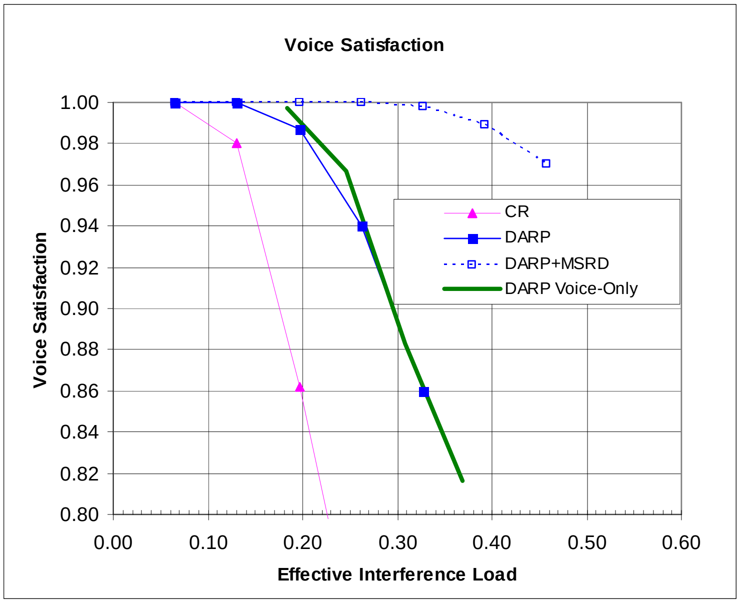

Figure 4 – Voice satisfaction versus Effective Interference Load, 50 km/hr

	Conventional Receiver	DARP	DARP+MSRD (2 dB AGI, 0.4 ρ)
System Voice Capacity (EFL at 98% FER <2%)	20%	32%	65%
Avg. User Throughput (bps) at 20% EFL	8,690	10,485	12,579

Table 1. System performance, flat 50 km/hr

4. Conclusions

In a mixed voice-data system, DARP increases per-user data throughput significantly, and increases voice capacity dramatically. The further addition of Mobile Station Receive Diversity (MSRD) greatly improves both voice and data. These results are consistent with [5], which noted the larger DARP gain for voice than for HTTP data. MSRD will benefit both the user and the operator, and should be considered favorably in the GERAN evolution.

5. Appendix

5.1. System Configuration and Parameters

Parameter	**Value**	**Unit**
Frequency	1900	MHz
Bandwidth	1.2	MHz
Reuse	1/1 (TCH)	-
Voice Codec	AMR 5.9 FR	-
Cell Radius	1000	m
Sectors (cells) per Site	3	-
Sector Antenna Pattern	UMTS 30.03	-
Propagation Model	UMTS 30.03	-
Log-Normal Fading: Standard Deviation	8	dB
Log-Normal Fading: Correlation Distance	110	m
Log-Normal Fading: Inter-Site Correlation	50	%
Adjacent Channel Interference Attenuation	18	dB
Handover Margin	3	dB
Antenna Gain Imbalance (AGI)	2.0	dB
Antenna Correlation (ρ)	0.4	-
Fast Fading	Flat	-
Mobile Speed	50	km/hr
Hopping	Random RF, uncorrelated fading	-

Table 2. System Assumptions and Parameters

5.2. HTTP Traffic Model

	Parameter	Value	Note
1	Session arrivals	Poisson	Mean 5 arrivals/hr
2	Number of packet calls in session	Geometric	Mean 5, max 15 (Re-cast OOR RVs)
3	Packet call size	Pareto	alpha = 1.1, k = 2.25 Kbytes, m = 225 Kbytes (Limit OOR RVs)
4	Packet call reading time	Geometric	Mean 5 s, no max
5	Packet size:	Semi-empirical	40% 40 bytes, 20% 576 bytes, 20% 1500 bytes, 20% Uniform (40-1500 bytes)
6	Number of packets in packet call	-	Depends on packet call size (RV) and packet size (RV)
7	Packet inter-arrival time	-	Depends on packet size and backhaul rate (32 kbps)
8	Data Erlangs/HTTP User	0.043	At MCS-2 (MCS-dependent)

Table 3. HTTP Traffic Model

6. References

[1] GP-040408, "A GPRS traffic model for SAIC performance evaluation", Nokia, 3GPP TSG GERAN #18, Reykjavik, Iceland, 2-6 February 2004

[2] Nokia, AHGEV-002, “MS Rx Diversity”, 3GPP GERAN Evolution Ad Hoc, Copenhagen, Denmark, 18-19 May 2005

[3] Motorola, GP-051504, “Observations on Receive Diversity Implementation and Performance”, 3GPP TSG GERAN #25, Montreal, Canada, 20-24 June 2005

[4] Motorola, GP-051979, “System Capacity of Mobile Station Receive Diversity With DARP”, 3GPP TSG GERAN #26,Schaumburg, IL, August 29 – September 2, 2005

[5] Nokia, GP-041057, “SAIC gains with mixed speech and GPRS traffic”, 3GPP TSG GERAN #19, Cancun, Mexico, 19-23 April 2004