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Unitary Beamforming Techniques Enhancing VoIP Call Admittance Capacity

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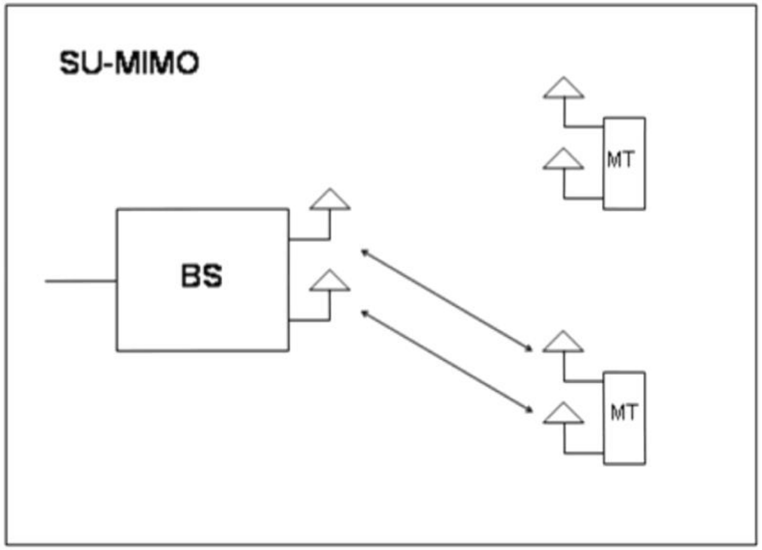
Introduction-Motivation

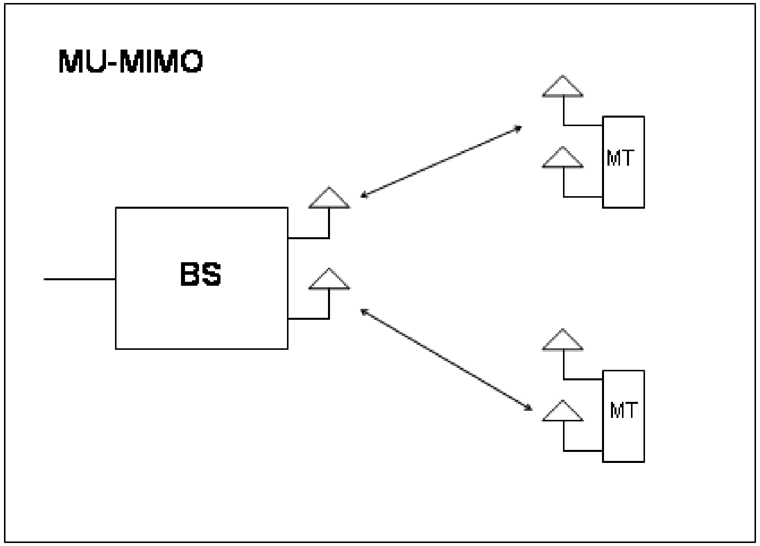
- Fourth Generation Networks will need to cope with increased data rate demands and improved QoS requirements
- MIMO-OFDMA identified as potential solutions
- 3GPP-LTE adopted unitary precoding techniques (SU-MIMO & MU-MIMO) for multiuser diversity exploitation
- This paper considers adaptive link and physical layer transmission techniques for multi-user MIMO systems, and resource allocation for wireless MIMO systems with QoS provision

Physical Layer Model

- Unitary codebook based beamforming is capable of achieving spatial multiuser diversity gain and spatial multiplexing gain
- Considerable lower overhead than the conventional eigen-beamforming approach
- Pre-defined set of precoding matrices applied to the transmitted signal.
- A delay oriented Link Adaptive (LA) scheme that incorporates an upper bound PER constraint (in order to avoid excessive delays arising from the retransmission of erroneous packets) is adopted

Physical Layer Model





- One MT is allocated across all spatial layers of the MIMO channel
- Aggregate channel strength information
- Loss in spatial diversity

- Different spatial streams assigned to different MTs
- Channel strength information across each stream
- Spatial Diversity Exploitation

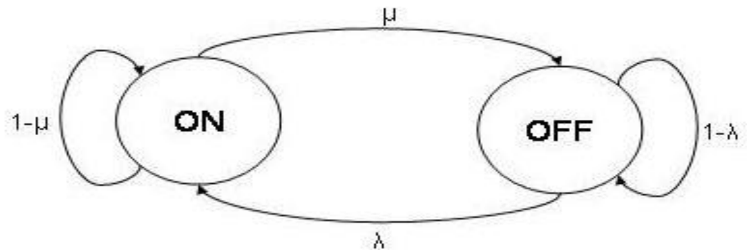
VoIP Traffic Model

- **Unsolicited Grant Service (UGS):**
  - Fixed amount of bandwidth for the duration of the call, no periodic bandwidth request or polling service
  - MAC overhead and uplink access delay can be minimised
  - Bandwidth Wasteful

- **Extended real-time variable rate (ERT-VR) with Voice Activity Detection (VAD)**
  - Resources are assigned for only the duration of talk spurts
  - The ERT-VR VoIP transmission with VAD modelled as a variable bit rate (VBR) system with active and silent periods

VoIP Traffic Model

- Two-state Markov process is used to model VoIP, representing the active voice and silent periods respectively



- Alternating periods of activity and silence exponentially distributed

Component	Distribution	Parameters
Active state duration $(1/\lambda)$	Exponential	Mean =0.4 s
Inactive state duration $(1/\mu)$	Exponential	Mean =0.6s
Packet Inter-arrival rate within a burst	Fixed	r = 5packet/s
Probability of transition from active to inactive state	N/A	μ=0.6
Probability of transition from inactive to active state	N/A	λ=0.4

PERFORMANCE ANALYSIS OF SU-MIMO and MU-MIMO

- OFDMA Parameters

Parameter	Value
FFT size	1024
Useful Subcarriers	768
Guard Interval Length	176
Subcarrier Frequency Spacing	10.94 KHz
Useful Symbol Duration	102.9 µS
MAC Frame Duration	5 ms
Channel coding	Punctured $1/2$ rate, convolutional code, constraint length 7, $\{133,171\}_{octal}$

- Transmission Modes

Mode	Modulation	Coding Rate	Coded Bits per OFDM symbol	Data Bits per OFDM symbol (*B*)
1	BPSK	½	768	384
2	QPSK	½	1536	768
3	QPSK	¾	1536	1152
4	16 QAM	½	3072	1536
5	16 QAM	¾	3072	2304
6	64 QAM	¾	4608	3456

- SCM urban micro path loss model for Line-of Sight (LOS) based on the Walfish-Ikegami-Model (COST-W1)
- 2x2 MIMO antenna configuration
- Link Throughput calculated according to $LT = B\left(1-\sum_{1}^{\infty}(PER)^{r}\right)R \approx B(1-PER)$

  B being the number of encoded data bits per OFDM symbol

  r the number of required re-transmissions

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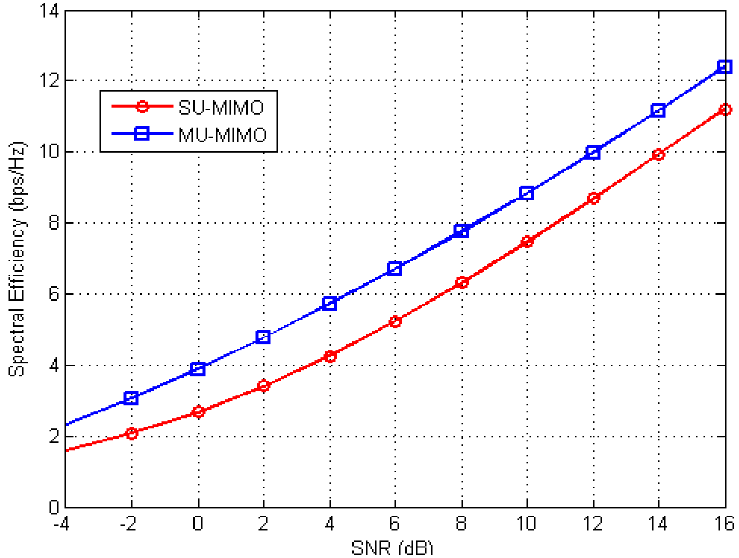
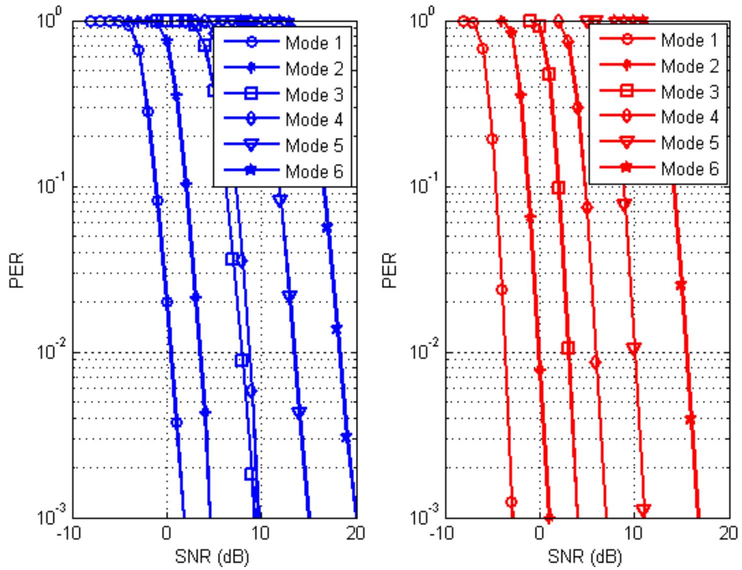
PERFORMANCE ANALYSIS OF SU-MIMO and MU-MIMO

- Theoretical Spectral Efficiency for SU-MIMO and MU-MIMO



- PER SU-MIMO-& MU-MIMO-OFDMA under different modes



- MU-MIMO achieves a higher data rate than SU-MIMO
- MU-MIMO tolerates lower SNR for the same PER requirement
- This can either be translated into a higher call admittance capacity, greater cell coverage or reduced power requirements

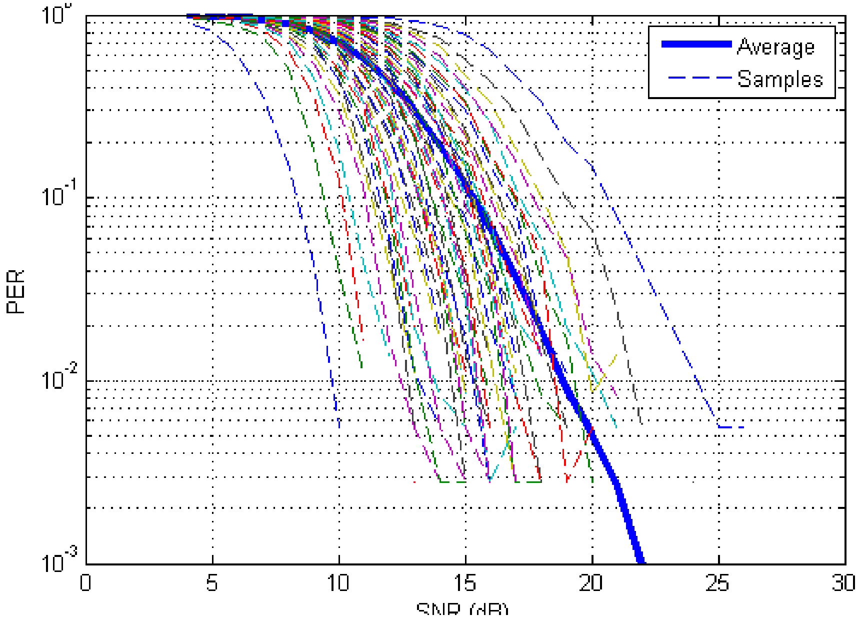
QoS PERFORMANCE WITH LINK ADAPTATION FOR VoIP

- Received signal quality in wireless systems is affected by various channel factors, e.g.
  - Distance of the MT from the serving BS
  - Path loss exponent
  - Log-normal shadowing
  - Fast Rayleigh fading and noise

- Systems that do not employ a Link Adaption (LA) strategy use a fixed transmission scheme designed to maintain acceptable performance for worst case conditions

QoS PERFORMANCE WITH LINK ADAPTATION FOR VoIP

- Wireless channels suffer from frequency selective fading
- The conventional notion of performing LA based on the average PER performance fails, since the stochastic nature of individual fading channel realisations can vary considerably from the average
- PER performance of a given fading realisation can vary by more than 10dB from the average



QoS PERFORMANCE WITH LINK ADAPTATION FOR VoIP

- Bit errors can be broadly attributed to the average received channel strength and the probability of resource allocation on fading instances
- An indication of resource allocation on fading instances, the ratio of the frequency channel responses with channel strength lower than the average received response is selected

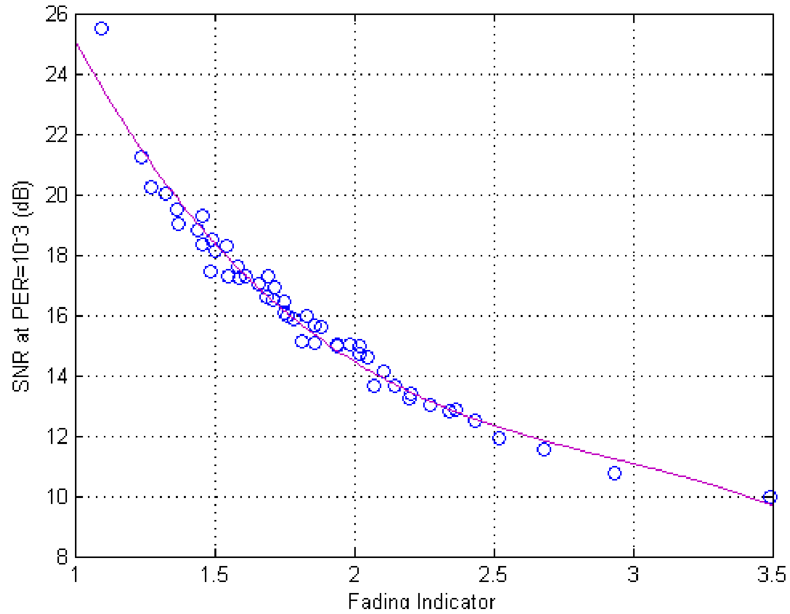
$$I_{fad} = \frac{\left|\overline{H}\right|^2}{\frac{1}{N_{SC}} \sum_{i}^{N_{SC}} F(i)}$$

where *F(i)=1 if* $|H_i| < |\overline{H}|$ *and F(i)=0 otherwise,*

$|H_i|$ *indicates channel strength on each frequency resource i*

QoS PERFORMANCE WITH LINK ADAPTATION FOR VoIP

- A high $|\overline{H}|$ value and a low number of deep fades describe a channel with a low likelihood of bit errors
- By mapping each fading realisation to a corresponding fading indicator value, a prediction on the required SNR for which a target PER requirement is met, can be made

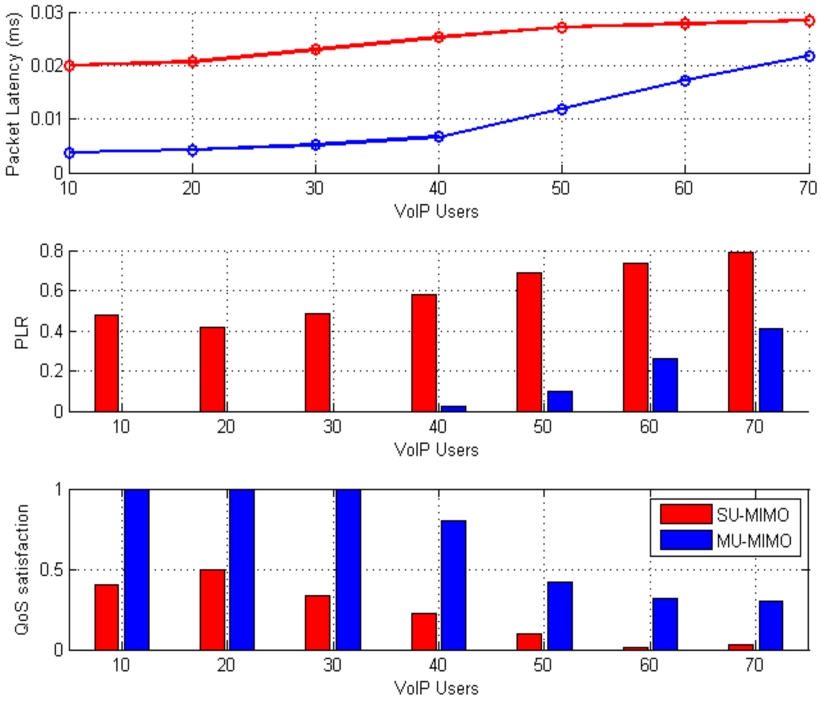


Simulation Results

- Max. tolerable packet delay: *30ms*
- Max. VoIP packet timeout ratio: *4%*
- Satisfied Customer Ratio: *The ratio of VoIP calls not exceeding max. packet timeout ratio over total number of calls*
- Users are randomly distributed within the cell, with a minimum range of 75m from the BS to the cell edge at 800m
- A BS transmit power: *43dBm*. Receiver Sensitivity:-120dBm.
- MU-MIMO provides significant benefits over SU-MIMO in terms of throughput
- The exploitation of spatial diversity can also improve the overall system fairness

Simulation Results: Packet Latency-Packet Loss Ratio (PLR)-Satisfied Customer Ratio

- MU-MIMO allows for lower delays due to increased number of available radio resources and enhanced resource allocation
- Correlation between packet latencies and PLRs. As the network becomes more congested, the resulting scarcity of available resources gives rise to increased buffer waiting times
- Maximum number of admitted VoIP calls with no QoS outage is significantly higher for MU-MIMO than for SU-MIMO



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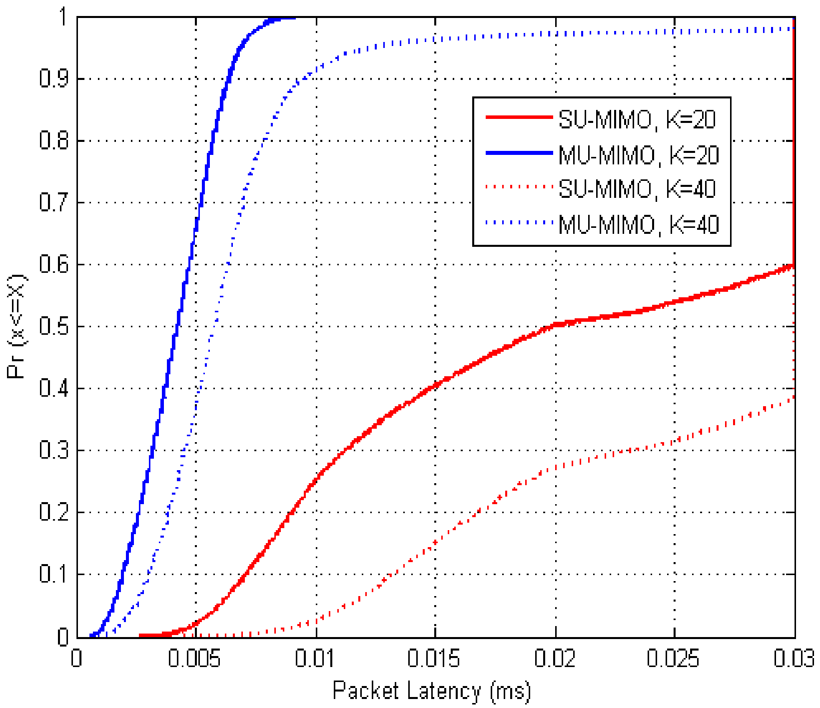
Simulation Results: Cumulative Distribution of Packet Delays

- CDF of packet delays arising from SU-MIMO and MU-MIMO for a total number of users, *K*=20 and *K*=40
- Average packet delay is directly related to the traffic load and the degree of diversity exploitation
- Fairer distribution of delays arising from MU-MIMO over SU-MIMO



Conclusions

- Examined QoS performance of two MIMO precoding schemes proposed for 3GPP-LTE, in a VoIP traffic system under a realistic channel environment
- A novel Link Adaptation strategy that considers the stochastic variability of fading channels has also been proposed
- Efficient exploitation of multiuser diversity across the spectral and spatial domain in form of MU-MIMO allows for notable increases at the call admittance capacity
- A more uniform distribution of packet delays across users experiencing distinctly diverse channel conditions has been observed
- These benefits have been attributed to the increased data rates and improved resource allocation, attained by MU-MIMO