Performance of TCP/IP Over Next Generation Broadband Wireless Access Networks

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Abstract

This paper presents performance evaluation results for Internet access over the next generation of broadband wireless access networks (2G-BWA). The challenge for these, mostly fixed access technologies stems from non line-of-sight deployment with high coverage and spectrum efficiency combined with link performance comparable to that of competing wired technologies (DSL, Cable modem). We present an overview of the features at the physical (PHY) and medium access control (MAC) layers which are key to a successful 2G-BWA design and demonstrate the impact of such features on end-to-end link performance with typical TCP/IP applications and wireless channel models.

Keywords

Broadband wireless access, MIMO, adaptive modulation, link layer ARQ protocol, TCP/IP.

1. Introduction

The demand for broadband Internet access is booming. Announced delays in the deployment of third generation high speed wireless networks (WCDMA and cdma2000 are unlikely to be widely available before 2003-2004), as well as slow progress in satisfying demand for wired solutions such as x-DSL and Cable modems place high expectations on alternative access technologies such as fixed wireless. Fixed wireless technologies of focus here address the lower frequency bands (MMDS, 3.5GHz) dedicated to mass market – residential, small offices home office (SoHo) use. Current internet broadband wireless access technologies are based on the existence of a line of sight link between the subscriber's unit and the access point (base station). This assumption allows to avoid multipath fading and equalization, however it puts very stringent limits on the scalability and ubiquity of the technology, preventing low cost mass deployment and also preventing any future evolution to support mobility. Figure 1 illustrates the main network components of BWA systems.

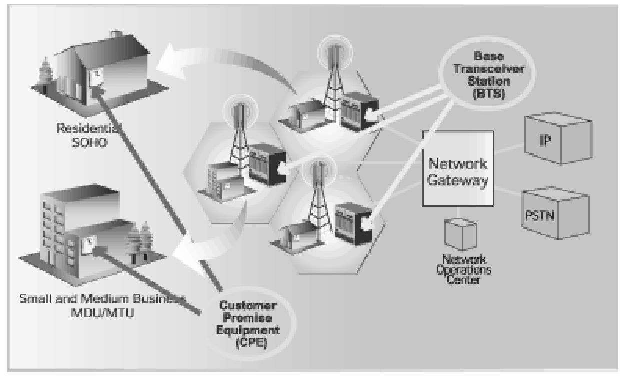

Figure 1. Illustrating BWA network components.

In order to successfully respond to the demand and carry their future share of access traffic, BWA systems must evolve into a second generation technology that can give high TCP/IP performance over severely fading channels caused by non line-of-sight deployment. The performance needs to be at least comparable to wired access: DSL and Cable modem. BWA systems need, above all, to make an efficient use of spectrum, while at the same time maintaining user friendliness especially during installation and low cost of operation. Typical overall figure-of-merit requirements for these upcoming systems are shown in Table 1.

Figure of Merit	Requirement
Aggregate Rates	4-7 Mb/s
Spectrum Efficiency	2 Bits/Hz/BTS
Coverage	4-6 miles (90% area)
Latency	Comparable to DSL
Link reliability	0.999

Table 1: Requirements for 2G BWA.

Non line-of-sight broadband wireless channels are prone to frequency selective and time selective fading, [1]. In addition, because of the quasi-stationary nature of the subscriber unit, the Doppler spread in the channel is

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low (less than 1Hz) causing fades to typically last longer than wireless mobile applications (up to a few seconds). Fades are due to moving reflecting objects in the environment such as tree leaves, cars and pedestrians, rather than the movements of the access unit itself. To provide good link performance at TCP/IP level over such channels, fade mitigation at either the PHY or MAC layers becomes absolutely critical. A number of emerging technologies that satisfy this need have been investigated for use in the AirBurst (a proprietary system being developed by Iospan Wireless Inc.) BWA system. Note that only generic or standard versions of the system features are mentioned here. Actual implementation details are system proprietary however. This paper first presents a tutorial background on the windowing and loss recovery mechanisms of TCP and address difficulties faced by TCP over unreliable and lossy wireless links. Then, a selection of technologies, both at the MAC and PHY, that are suited for non line-of-sight BWA are presented. Next, end-to-end link performance (including TCP/IP) for a typical 2G-BWA system exposed to time/frequency fading channels is evaluated. In particular, the focus is on the respective impact of key PHY and MAC layer features on end-to-end TCP/IP performance over these channels. We conclude with design recommendations for 2G-BWA.

2. TCP Slow Start and Congestion Avoidance

In this section, we recall the TCP windowing and loss recovery mechanisms and emphasize differences between several widely used versions: Tahoe, Reno and NewReno.

TCP is a dynamic window based flow control protocol. The window size, denoted by $W(t)$, varies in response to packet acknowledgments (ACKs) and loss detection. While the receive processes are the same for all TCP versions, their transmit processes and window adjustment algorithm in response to loss detection are different [2].

At any given time, *t*, the TCP transmitter maintains several variables for each connection: lower window edge, congestion window and slow start threshold. *Lower window edge* $X(t)$, denotes that all data packets numbered up to $X(t) - 1$ have been transmitted and acknowledged. Next eligible packet the transmitter can send is numbered $X(t)$. Each ACK receipt causes $X(t)$ to advance by the amount of acknowledged data and, thus, lower edge window evolution is monotonically non-decreasing time function. The transmitter's *congestion window*, $W(t)$, defines the maximum amount of data packets the transmitter is permitted to send, starting from $X(t)$. Control mechanism upon ACK reception allows $W(t)$ to increase or decrease within W_{max} boundaries. During connection set up, the receiver advertises a maximum window size, W_{max}, so that the transmitter does not advance instantaneous window $W(t)$ beyond this value during whole connection period. *Slow start threshold* $W_{th}(t)$ controls the increments in $W(t)$. If $W(t) < W_{th}(t)$ transmission is assumed to be in slow start phase, where each ACK causes $W(t)$ to be incremented by 1. This rapid growth switches to slow increase, once the window crosses slow start threshold, that is, when $W(t)>=Wth(t)$. This phase, called congestion avoidance phase, aims to cautiously probe for extra bandwidth. Each ACK reception increments window size $W(t)$ by $1/W(t)$.

On the receiver side, packets can be accepted out of order, but will be delivered only in sequence to intended TCP user. The destination returns *cumulative* acknowledgement for every correctly received packet. Each ACK carries next in-order packet sequence number expected at the receiver, which is the first among the packets required to complete the in-sequence delivery. Thus, a single packet loss can be detected if the same "next expected" number is received in all consecutive acknowledgements of packets successfully transmitted after the lost packet. Reaction on packet loss detection depends upon the TCP version. In the simplest case, TCP OldTahoe, the TCP transmitter continuously sends packets till the congestion window is exhausted and then waits for timer expiry. The sender maintains a timer only for the last transmitted packet, that is, each time a new packet is sent, the transmitter resets the already running transmission timer. The timer is set for a round trip timeout *(rto)* value that is derived from a smoothed round-trip time *(rtt)* estimator applying the low pass filter over the last estimate and the current measurement of the *rtt* [3]. The timeout values are set only in multiples of a timer granularity: in a typical implementation the timer granularity is 500ms. According to the basic recovery algorithm, upon timeout window parameters are adjusted - $W_{th}(t+)$ is set to $W(t)/2$ and $W(t)$ is reset to 1, and retransmission is initiated from the first missing packet.

More sophisticated TCP protocols such as Tahoe and Reno, implement *fast retransmit* and *loss recovery* procedures. The Fast retransmit procedure takes advantage of additional information that duplicate acknowledgements carry. The transmitter waits for several duplicate acks to arrive (typically three) to account for possible network delay and packet reordering. Then, the missing packet is retransmitted before the timer expiry. In subsequent loss recovery, TCP Tahoe transmitter behaves as if a timeout has occurred and resorts to the basic recovery algorithm.

In the case of Reno, *only* the lost packet is fast retransmitted and loss recovery procedure is handled differently. Congestion window recorded at this time is referred as *loss window*. After N^{th} duplicate ACK at timet, sender adjusts slow start threshold $W_{th}(t^+)$ to $W(t)/2$ and $W(t^+)$ to $W_{th}(t^+) + N$. Congestion window $W(t)$ is additionally incremented by N to account for packets that have successfully left the network and produced duplicate ACKs. While waiting for the ACK for the first retransmission, any successfully received outstanding packet produces duplicate ACK, which further increments $W(t)$ by 1. If single packet were lost, the packet retransmission would complete loss recovery. At this time, congestion window $W(t)$ is set to $W_{th}(t)$, and the transmission resumes according to the normal window control algorithm. However, the algorithm can experience a possible stall time if multiple packet losses occur within the loss window. In this case, the congestion window might close before sufficient number of duplicate ACKs has been generated to trigger multiple fast retransmits. In that case algorithm waits for timer expiry. The Reno recovery algorithm outperforms Tahoe version, but is pessimistic and suffers performance degradation in case of multiple losses within the loss window [2].

New Reno attempts to fix multiple loss drawback by setting fast retransmit and congestion window adaptation as in Reno and implementing the loss recovery mechanism as in Tahoe. In this paper, we focus mainly on TCP Reno.

2.1. Performance of TCP on wireless links

From the discussion above we see that TCP is not designed for lossy links. The TCP protocol has been developed for wired networks to deal with network congestion, rather than non-negligible random losses. On the contrary, wireless link impairments raise the packet error probability several orders of magnitude. The TCP sending rate drops upon loss detection. Afterwards, TCP loss recovery kicks in and offered throughput increases gradually. Congestion window increase, clocked by arrival of acknowledgements, largely depends on round trip time. In other words, if round trip time is large and the congestion window decreased frequently due to multiple losses, the TCP connection may not be able to fully utilize available channel bandwidth. Although some of the previous studies show that different versions of TCP protocol perform better or worse in response to wireless lossy link with correlated errors, none of them gives satisfactory performance. This calls for new techniques that will allow reliable transmission in heterogeneous networks comprising both wired and wireless links. Comparative analysis of several higher layer schemes proposed to alleviate the effects of non-congestion related losses has been conducted in [4]. The authors classify the schemes in two categories according to the approach they take to improve TCP performance in lossy systems. The first category includes link-layer protocols, split connection approaches and TCP aware link layer schemes. These protocols try to locally solve the problem and hide wireless losses from the TCP sender. As a result, wireless hops appear as high quality links with reduced effective bandwidth. The second class of techniques attempts to make the sender aware of the existence of non-congestion related losses. The idea is that sender restricts itself from invoking congestion avoidance algorithm in response to such events. However, second class schemes require undesirable change in TCP end-to-end algorithm. Comparison analysis shows that local protocols outperform the second class techniques, [4]. Moreover, reliable link layer schemes offer higher throughput than split connection approach, while at the same time preserve end-to-end protocol semantics.

3. Design features for 2G-BWA networks

In this section, the focus is on BWA system design features that attempt to overcome unreliability introduced by time and frequency selective fading and interference-prone channels. Both PHY and MAC layer features are presented and TCP fixes that have been proposed to deal with wireless channel behavior are ignored. The focus here is on a set of *selected* features that the authors view as particularly critical or relevant. The MAC layer ensures a low resultant error rate seen by TCP. The PHY layer improves link quality and enables higher data rates.

3.1. MAC Layer

Automatic Retransmission/Fragmentation (ARQ/F)

ARQ/F is widely considered a key tool to deal with bursty errors occurring in wireless channels. A low-latency acknowledgment and retransmission mechanism is implemented at the MAC layer between the subscriber unit (or CPE) and the BTS. The MAC layer fragments IP packets into 'atomic' data units (ADU). Only lost ADUs are retransmitted. A technique complementary to coding, ARQ/F only introduces redundancy during the fraction of time when data gets corrupted. A finite buffer makes ARQ/F efficient in dealing with short as well as long fades. ARQ/F removes packet errors at the cost of only moderate additional link latency. ARQ/F-based systems can be designed to operate at high BER levels while still providing satisfactory TCP/IP performance, as shown later in the results section.

At a transmitter, PDUs received for transmission from higher protocol layers are fragmented into ADUs and stored in a transmit buffer. These ADUs are normally transmitted sequentially, with each outgoing ADU being uniquely identified by its sequence number. Each ADU is kept stored in the transmit buffer even after it has been transmitted. At a receiver, uncorrupted incoming ADUs are stored in a reassembly buffer. Whenever all of the ADUs comprising a PDU are received, the reassembled PDU is passed up to the higher layer protocols. However, there may be situations where one or more ADUs comprising a PDU is received corrupted and needs to be retransmitted. In order to notify the transmitter about these corrupted ADUs, a receiver periodically generates an Acknowledgement message (ACK). This message enumerates the sequence numbers of ADUs that are in the receiver's reassembly buffers. When a transmitter receives an ACK, it retransmits the "missing" ADUs from its transmit buffers. These retransmitted ADUs are given higher priority over regular ADU transmissions. ADUs that have been successfully received at the receiver are freed from the transmit buffer.

The ARQ protocol is based on a sliding window mechanism, that is, each flow is allowed to transmit only if it has no more than a window size (*W*) worth of unacknowledged ADUs. More formally, the following inequality must always hold:

$$T(N) - T(U) <= W \text{ if } T(N) >= T(U) \qquad (1)$$

$$MAX\_SEQ\_NUM - T(U) + T(N) <= W \text{ if } T(N) < T(U) \qquad (2)$$

where *MAX_SEQ_NUM* denotes the size of the sequence number space, *T(N)* denotes the sequence number of the next in-sequence RLC ADU eligible to be transmitted at a transmitter and *T(U)* denotes the sequence number of the "oldest" RLC ADU that has not been positively acknowledged by the receiver. The window size W shall be configurable at the time of link initialization but it can be no larger than *MAX_SEQ_NUM*/2 ADUs. This limit arises from the size of the sequence number space.

Weighted Round Robin Scheduling

At the beginning of every frame, the BTS transmits a MAP message that allows each NAU to determine whether it is supposed to receive or transmit ADUs during that frame.

For a downlink or uplink transmission, each MAP allows a particular NAU to determine which time slots it should decode or transmit in, respectively, and what modulation/coding is currently being used for the ADUs.

The scheduling policy used to compute the uplink and downlink transmission schedules at the BTS has to meet the following requirements and constraints:

- Ensure that CBR service flows get allocated a constant bandwidth (with acceptable jitter)
- Ensure that UBR service flows share available bandwidth in a fair manner
- Support spatial multiplexing
- Support channel dependent scheduling. That is, compute transmission schedule based on the knowledge of modulation/coding being used over current channel state
- Provide ability to temporarily shut off transmission to a NAU with a bad link

Adaptive Modulation (AM)

Adaptive modulation and coding lets a user adapt its data rate as a function of channel conditions (e.g. SINR) and has been popularized in the EDGE cellular standard [5]. AM allows several times rate improvement by exploiting all margins of signal to noise ratio available at any time/location. The idea behind location adaptive modulation is depicted at Figure 2. By comparison, a non-adaptive system is deployed with most conservative modulation/coding for all users in order to preserve good coverage and frequency reuse. Note that self-adaptivity (to even slowly changing conditions) by the CPE at one location is a key requirement to extract significant capacity gain. Additional gain is obtained by adapting faster. For TCP/IP, AM will result in larger pipe size and

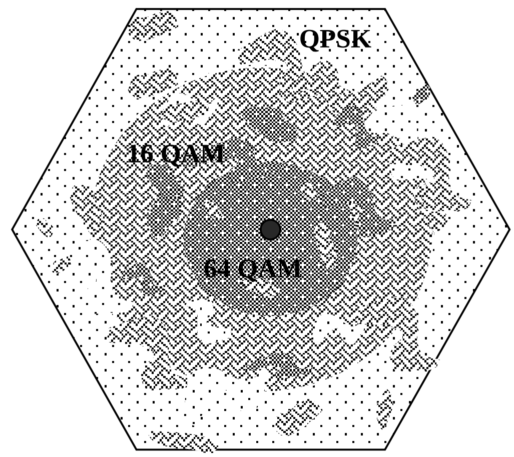

Figure 2. Adaptive Modulation.

additional statistical multiplexing. To detect an appropriate modulation/coding level, the MAC layer uses a set of statistics obtained from the PHY layer. The set of statistics involves a combination of Bit Error Rate, Packet Error Rate and signal to noise ratio. The details are not disclosed in this paper. However, it is assumed that each user in the cell is assigned a modulation and coding level that is a function of individual channel conditions and that guarantees a given target Packet Error Rate.

3.2. Physical Layer

Spatial Diversity (SD)

SD is obtained through the use of multi-element antennas at either the BTS and/or the CPE. Antenna combining is used to deal with fading, Figure 3, and allows advanced interference canceling algorithms. The basic idea of diversity combining is that several uncorrelated fading elements are much less likely to fade simultaneously than a single element. Use of SD can reduce the SNR requirement by 10-15 dB with no loss of TCP/IP performance. This is exploited to extend the coverage, increase data rates, and most importantly to allow a tight frequency reuse that will work in arbitrary terrain (flat or not). Recent measurement campaigns for BWA suggest that decorrelation is achieved with 1-2 wavelengths of element separation; less spacing is required with dual-polarized antenna elements [6].

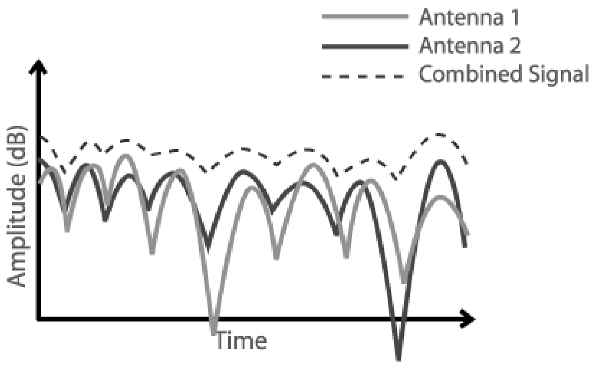

Figure 3. Multiple antennas provide a robust source of diversity in Non LOS channels.

The signal processing technique used to combine the antennas can vary. Maximum Ratio Combining (MRC), Minimum Mean Square Error (MMSE) combining are typical examples of combining algorithms when the channel coefficients are known (e.g. at the receiver) [7]. On transmission, where the exact channel coefficients cannot be known unless a bandwidth consuming feedback link is used, a blind transmit spatial diversity must be used. Examples are space-time coding algorithms recently developed [8]. Delay-diversity is used for transmitting and MRC is used for receiving.

MIMO and Spatial Multiplexing (SM)

In the case of multi-element antennas at both ends (multiple-input multiple-output or 'MIMO'), an additional data rate increase can be obtained by exploiting simultaneous transmission over the eigenmodes of the channel as seen in [7] and [9]. A simple spatial multiplexing algorithm works as follows (more advanced variations combined with coding are typically used in practice): a high rate signal is demultiplexed into set of independent bit streams which are independently transmitted by the multiple antennas. The signals are mixed with each other in the channel since they occupy the same time and frequency resource. At the receiver, the multiple antennas are used to first learn the spatial signature (channel) of each stream using training sequences then combined so as to separate the different bit streams from each other. The streams are finally multiplexed to offer the original high rate signal. This operation consumes only the bandwidth normally needed to transmit one single bit stream. The increase in data rate is achieved at the expense of increased interference. The procedure is illustrated in Figure 4.

When there is only one usable eigenmode due to rank deficiency of the MIMO channel, the streams are not separable and the system must fall back to a diversity scheme [9], using proprietary adaptation algorithm. This technology is particularly well suited for BWA because,

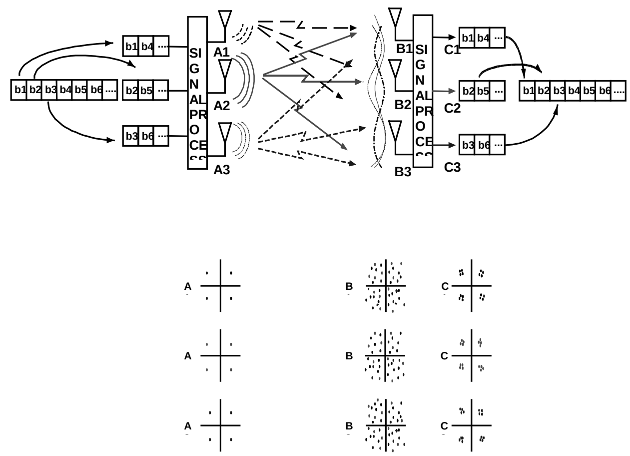

Figure 4. Simplified Spatial Multiplexing scheme with three antennas. This gives three fold data rate improvement.

unlike handsets, compact multi-element subscriber units can be designed with 2 or 3 elements. The gain in data rate is proportional to the minimum of the number of transmit and receive antennas. The impact of this technology on TCP/IP performance is also examined here.

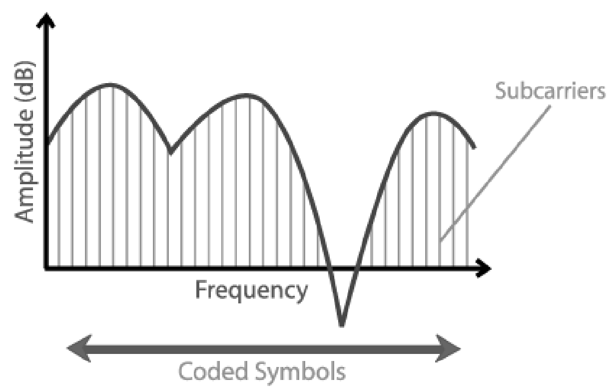

Figure 5. OFDM with coding transforms multipath into a source of anti-fading diversity.

Frequency Diversity

Broadband transmission over multipath channels introduces frequency selective fading, as illustrated in Figure 5. Mechanisms that spread information bits over the entire signal band will mitigate the effect of fading occurring at certain frequencies. An example of such multipath-friendly mechanisms is frequency-coded multicarrier (e.g. OFDM) modulation [10]. This technique requires lower SNR to achieve the same TCP/IP performance.

4. End-to-end performance evaluation

4.1. Simulation Methodology

The main objective of end-to-end simulator is to capture all aspects of system design and examine their impact on TCP/IP. The simulations are organized into several hierarchical layers, where each module sees the abstracted behavior of a layer below. The simulation is constrained from the bottom by BWA channel models and from the top by TCP/IP usage models. The three main design layers with several degrees of abstraction and implementation detail are captured, namely Transport, MAC and PHY layers. Different tools are used to implement each layer.

Network Simulator (NS) [11] is used to capture user behavior and TCP-based transport. Detailed MAC layer functionality is developed in C/C++ and interfaced to NS. PHY layer behavior is abstracted through error and data rate traces. These traces are obtained through several levels of PHY simulation and reflect performance of coding, modulation and multiple antennae processing techniques over realistic fixed broadband wireless channels. Realistic channel models are developed from fixed MIMO channel measurements.

The NS module models one target multi-user TDMA-based cell. However, the interference caused by other cells is also captured through the data and error rate traces, assuming a 2x3 frequency reuse. Wired backbone network is represented as a set of web servers (traffic sources) connected to the BTS through high bandwidth, high propagation delay and low BER links. The wired links converge to the BTS (Figure 6) which maintains separate queues for each network access unit (NAU). Multiple TCP connections, originating from different severs can get routed to the same queue modeling a business customer or residential users with multiple opened connections. Packets are transmitted over the air and are subject to errors as predicted by PHY simulations. A connection from the BTS to any particular NAU is represented with a link whose average quality is dictated by long-term channel condition specific for each subscriber location.

4.2. MAC Model

The MAC layer model captures detailed design features such as segmentation and re-assembly of IP packets, scheduling, ARQ, UL access via contention and link adaptation.

Received IP packets are fragmented into atomic data units (ADUs) and resulting smaller packets are stored for transmission. Receive module ensures in-order delivery to higher (TCP) layers. An ADU scheduled for transmission can be lost with a certain probability determined by the instantaneous channel state and transmit mode. Lost ADUs are identified and marked for retransmission. The scheduler enforces a weighted round robin policy. Allocations are made for a number of frequency blocks considering transmit mode and quality of service requirement of a particular user. On reverse link, from NAU to BTS, initial access is obtained through a contention channel. Once data transfer starts, additional bandwidth requests are piggybacked on the data ADUs. If enabled, both UL and DL are subject to transmit mode adaptation based on SNR measurements made at the receiver.

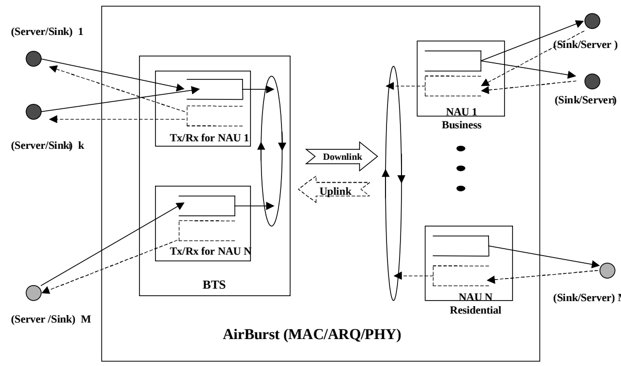

Figure 6. Simulation Environment

4.3. PHY Model

The PHY performance is summarized through estimation of modem setpoints for each transmit mode. A transmit mode is comprised of a coding mode (choice of MQAM and coding rate) and antenna mode (diversity or spatial multiplexing). The modem setpoint for a particular mode, as defined by the modulation, coding and number of transmit antennas, is defined as the signal-to-noise ratio (SNR) required to achieve a desired pre-ARQ operating error rate. Here, two cases can be distinguished. If coding mode were static over time, modem setpoint would represent the average SNR level over a long period of time attained at one receive antenna element. In the second case, link adaptation tracks channel variations, allowing maximization of data rate whenever possible through selection of 'optimal' transmit mode. In this case both packet error rate trace, as well as corresponding mode trace are generated and supplied as input to the end-to-end simulator.

4.4. Wireless Channel Model

Physical properties of wireless communication channels exhibit lots of impairments unknown to their 'wired' counterparts. In a wireless system, a signal transmitted into the channel interacts with the environment in many complex ways. Signal propagation is subject to scattering, reflection, refraction or diffraction of surrounding objects, such as buildings, hills, streets, trees and moving mobiles, as shown on Figure 7. The received signal is much weaker due to wireless propagation impairments such as path loss, shadowing and fast fading. Path loss is the drop in radiated signal power due to the distance from the transmitter to the receiver, environment, water, foliage, etc. Ideal free space propagation follows inverse square law spreading with the distance, whereas in real cellular environment path loss exponent varies from 2.5 to 5.

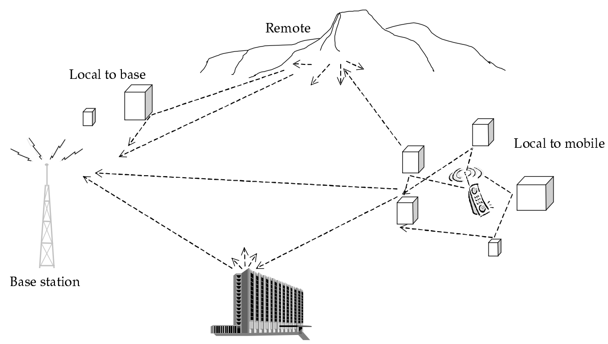

Figure 7. Radio wave propagation with diffraction, refraction and multiple reflection

Shadowing, or slow fading, comes from signal blocking by large obstructions such as buildings and hills, that are positioned between base station and access terminal. Multipath fading plays a central role in determining the nature of wireless channel. The result of signal interactions with numerous physical objects in its path to the receiver is a presence of many signal components, or multipath signals, at the receiver. Multipath arrives from different directions, with different attenuation and propagation delays resulting in received signal amplitude that varies in many dimensions – time, frequency, space.

Wireless channel characteristics are measured through several key parameters: Doppler spread, delay spread and angle spread.

Time selectivity, measured through Doppler spread, comes from the fact that pure a tone is frequency dispersed due to environment change in the receiver's vicinity. In the mobile channel, received signal envelope has Rayleigh distribution in the time domain, which translates into Doppler power spectrum of the received signal that has "horn" shape. On the other hand, fading characteristics of the fixed wireless channels are very different from mobile case and result in "rounded" Doppler spectrum, [12]. Techniques that exploit this property are adaptive modulation (AM) that tracks channel quality over time and radio link protocols, like ARQ/F, that attempts to fix fades by retransmission of lost packets in good channel states.

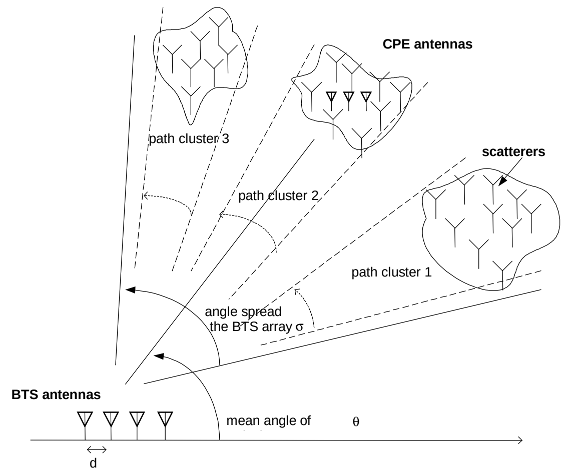

Figure 8. MIMO system channel model

Frequency selectivity or multipath delay spread is caused by several distinct and delayed versions of transmitted signal. Frequency selective implies that channel response depends upon the frequency. It is characterized by coherence bandwidth representing maximum frequency separation for which channel response remains correlated. Signal components delayed by even a small fraction of symbol period gives rise to inter symbol interference. OFDM is well suited to combat frequency selectivity either by coding over sub-carriers or adaptively loading sub-carriers with high SINR.

Space selective fading, or angle spread refers to distinct multipath angle of arrival that translates into signal amplitude that depends on antenna location. This property allows for space and MIMO diversity techniques in the systems with antenna arrays.

Depending on the relationship between signal parameters (bandwidth, symbol period) and channel parameters (delay spread, Doppler spread) signal distortions can be significant or on the contrary negligible. The level of detail about the environment a channel model must provide is highly dependent upon the radio architecture. Propagation model parameters depend upon terrain, wind speed, season/ time of the year, traffic density and proximity, height and beam width of transmit and receive antennas, as well as type of system under consideration. To predict the performance of single antenna narrow-band system, received signal power and/or time-varying amplitude distribution may suffice. However, for emerging wideband multiple antenna arrays information regarding the signal multipath and angle of arrival structure is needed. In other words, wideband MIMO channel is properly characterized by space-time channel model, as represented on Figure 8.

The channel models used in simulations reflect specific broadband wireless channels developed from measurements and published by Sprint and Stanford University [12]. These models capture radio link impairments mentioned above. We emphasize the use of non-line-of-site (NLOS) models with low Ricean K factors in order to design system with high reliability requirements. The use of multiple transmit and receive antenna calls for antenna correlation modeling. The system under study assumes cross-polarized antennas. Cross-polarization gives rise to attenuation due to polarization mismatch (XPD) between transmit and differently polarized receive antenna. For LOS component we use 10dB attenuation, but for Raleigh component XPD loss goes as low as 4dB. This effect is included in PHY modeling. Key parameters for the different channel models used in the analysis, SU1-SU6, are listed in Table 2.

	SU1	**SU2**	**SU3**	**SU4**	**SU5**	**SU6**
rms delay spread	0.1	0.2	0.3	1.3	3	5.2
overall K (linear)	10	5	0	0	0	0
antenna correlation	0.7	0.5	0.25	0.25	0.25	0.25

Table 2: Key parameters for channel models

4.5. Traffic Model

The exponential growth of World Wide Web users causes the traffic originating from Web applications to dominate the Internet both in terms of volume and active users. This trend, likely to continue in next years causes downstream and upstream link asymmetry. Majority of the Internet applications today, such as web-browsing and file downloads, require larger bandwidth pipes on the downstream. However, there is an increasing demand for streaming data (audio and video) applications, video conferencing and high quality of service (QoS) services that will require networks to support more symmetric traffic.

Accurate traffic modeling has high relevance in system evaluation and capacity planning. In particular, we are interested to model traffic requests that mimic a certain population of real users. Literature shows that such model has been difficult to develop due to a number of unusual Web workload characteristics, [13]. Empirical studies of operating servers show that their workload is highly variable, both in volume and number of open connections. This feature indicates that proper attention should be paid to properties of Web streams such as request inter-arrivals and file sizes. An important Web traffic characteristic is self-similarity, that is, significant variability over a wide range of time scales. Self-similarity negatively impacts network performance and becomes important to address.

In our system evaluation we focus on a traffic model that attempts to closely imitate HTTP requests originating from a population of Web users. We work with a traffic model characterized at two levels: macroscopic and microscopic.

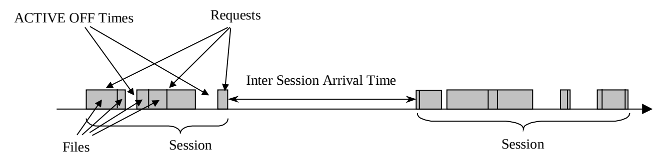

Figure 9. User activity model

The macroscopic description determines the business to residential user penetration ratio, the number of active connections per business NAU, and NAU activity factors across the subscriber population. In the simulations we assume that a set is composed of 20% business NAUs with 5 active connections each.

The microscopic description describes individual user behavior over time. A user can request data or act as a server to a remote client. We assume data transfers on the downlink have an average workload of 20 Kbps per connection and 4 Kbps on uplink. Introduction of uplink traffic diminishes link asymmetry and allows for more symmetric loads that account for future traffic development. In addition to these data workloads, control traffic in terms of MAC/ARQ or TCP acknowledgements is inherently generated. The workload is averaged over several www sessions, including 'think' time.

According to the standardized SURGE www usage model [13], each user starts several *sessions* during an online period. User behavior during a session is described as a superposition of *active downloads* and *think times*. Tails of the download (or request) sizes are described by a Pareto distribution, with the body of download sizes following a lognormal distribution. The average request size is 27 KBytes. The *think time* is Pareto distributed with an average of 3.5 seconds. The model parameters are shown in Table 3 and illustrated in Figure 9.

Request Body Log-normal	Request Tail Pareto	Active OFF Pareto	Session Length Inverse Gaussian
$\mu = 7.881$ $\sigma = 1.339$	K = 34 α=1.177	K=1 α=1.4	$\mu = 3.86$ $\sigma = 6.08$

Table 3: Parameters of www traffic usage pattern and file sizes. The average request size is 27kB.

5. Simulation Results

Next, simulation results to estimate the performance benefits of key aspects of system design that enable efficient broadband wireless operation are presented. Here we specifically address adaptive modulation (AM), space diversity (SD) and MIMO technique, operating error rate, impact of Doppler spread and give an example of TCP response to wireless channel.

Adaptive modulation and coding allow the system to adapt transmit modes across users in a cell, as a function of channel conditions at each location. The highest AM allowable rate is 12 times that of non-adaptive system deployed with the most conservative mode to preserve required coverage and frequency reuse. Diversity of transmit modes is obtained through MIMO and SD features of physical layer. On Figure 10, performance of both systems as a function of offered load is presented. We observe that non-adaptive system sustains several times lower workloads with even worse request service delays. In other words, good coverage of non-adaptive system is bounded by quality of service requirement, which in return greatly reduces number of users the system can support.

From Figure 10 we also observe how unequal data rate for users of disparite SNR levels reduces system's offered troughput. Suppose we separate users into N classes according to their SNR levels and corresponding supportable rate levels. User in class n receives data at rate R_n bits per second, where $n=1,2\ldots N$. Also suppose that all users request equal traffic amounts. This is true since all users are assumed to produce statisticaly the same traffic traces.Then, relative frequency of packets of class n, denoted by P_n, is proportional to the number of users in the class. Assume infinite traffic load of each class and suppose a scheduling scheme where slots are assigned one at a time succesively to each user.

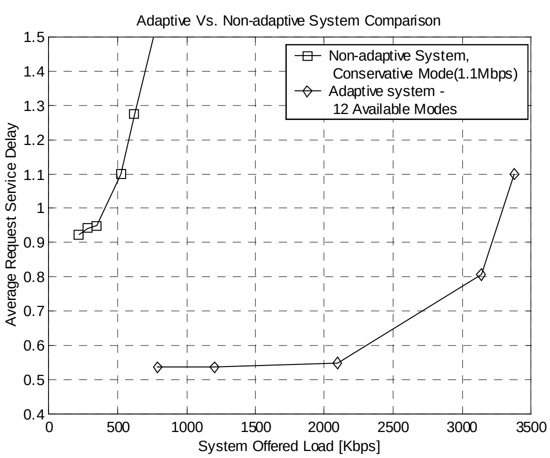

Figure 10. Adaptive modulation offers several fold improvement in capacity, efficiency and QoS. Note that the offered load does not include ~16% of upper layer protocol overhead.

In other words, bandwidth share of a given class is proportional to the class size. Then the system's avarage data rate, which we define as *throughput* is

$$R_{av} = \sum_{i=1}^{N} P_n R_n \text{ b/s} \qquad (3)$$

This means that fixed size packets of lower data rate users will have proportionally higher latencies. That is, if latency of a B-bit packet is considered, the number of slots consumed by each user in class n will be B/R_n, and hance the latency L_n is inversely proportional to rate R_n.

Suppose on the other hand that we require that all users have essentially the same latency irrespective of the R_n they can support and that finite amount of traffic is requested by each user within certain time window. Then, each class will consume bandwidth directly proportional to its size *and* inversly proportional to its rate. Round Robin scheduler assumption and QoS constraint on bounded delays require system to operate below its full capacity. Achieved system throughput differs from the average data rate, presented in equation (3), due to multiplexing of users with disparate SNR levels. Assume that each user requests B bits and that users of n-th class require $S_n = k/R_n$, k being a constant, slots to trasmit those bits. In this case we define *effective throughput* as :

$$R_n = \frac{\sum_{n=1}^{N} P_n R_n S_n}{\sum_{n=1}^{N} P_n S_n} = \frac{1}{\sum_{n=1}^{N} P_n / R_n} \, b/s \qquad (4)$$

For example, SNR distribution across the cell derived in AirBurst system with 5% average ADU error rate, yields an average *throughput* of 6 Mbps which translates into 4 Mbps of *effective throughput*. From Figure 10, we see that after including 16% of upper layer protocol overhead, system offered load reaches the maximum of 4 Mbps of effective user traffic at close to 100% link utilization.

Above analysis indicates cost paid in throughput for latency equalization. Allocation which is less bandwith "unfair" can be made if delay garantees are relaxed for the user with worse link conditions.

MIMO and SD diversity are key to enhancing system performance. They take advantage of rich fading structure of wireless channel and allow multiple transmit modes to be deployed. To achieve the same operating target error rate, physical layer simulations estimate that a user in a single antenna system (SISO) requires 12 dB additional SINR margin compared to MIMO system with 2 transmit and 3 receive elements. System analysis translates this margin into poor coverage of SISO deployment so that attainable cell size in a macrocell environment shrinks down 2.5 times. To preserve the same large cell size, a cell edge user in SISO system would need to operate at 50% packet error rate, unacceptably degrading post-ARQ delays. Furthermore, adaptive modulation can not exploit higher modes due to stringent SINR requirement. While preserving the target error rate for a given cell, average downlink peak data rate drops almost 3 times in SISO system.

Given above-mentioned underlying physical layer differences, TCP level quality of service is compared on Figure 11. The cumulative distribution function of throughput during file download is presented for files above 64KB. We exclude “small” files from the plot, since measured throughput for "small" files is gated by the TCP rather than PHY and MAC layers. In particular, connection round-trip time and slow start phase of congestion avoidance algorithm dictate the rate at which packets arrive at the wireless link. "Small" files are never able to fully utilize available bandwidth because the transmit window doesn't advance into congestion avoidance phase. Also, large round trip times cause large delays between consecutive congestion window increments, due to window advancement clocked by ACK arrivals. This effect is also present during a “big” file download, but only for a fraction of actual download time. Thus, the achievable throughput *during file downloads* asymptotically approaches *effective throughput* because of TCP.

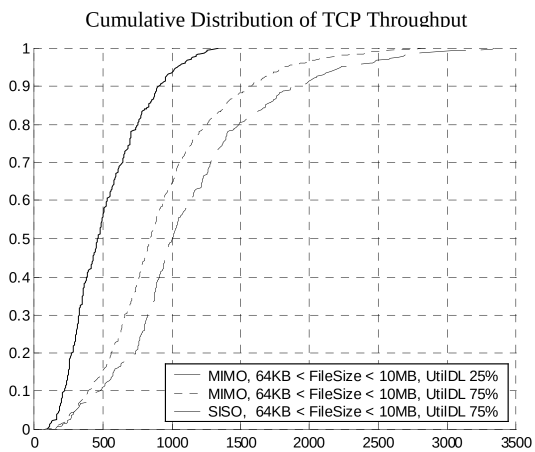

Figure 11. MIMO system offeres gain in coverage, thoughput and increases load the system can support

The number of users that build link utilization to 75% in SISO, only cause 25% link utilization in MIMO deployments. This corresponds to a three-fold difference in average data rate in these systems. In other words, to bring utilization back to 75%, the MIMO system can support up to three times more active users per cell site, while at the same time offering better quality of service.

Operating ADU error rate plays a pivotal role in user perceived quality of service and area coverage. Table 3 lists the peak and effective data rates averaged across the cell and coverage gain for several target error rates under study. The results presented are for a macro-cell environment defined by cell radius of 6 miles, rooftop NAUs and a BTS antenna height of 30m. Two interesting trends can be derived from the data in Table 4. Firstly, the average gross rates improve as the ADU error rate increases, due to the fact that minimum SINR requirement for a given mode (modem set point) is relaxed and allows users with poorer link quality to take advantage of higher modes. The rationale is to let ARQ recover from deep fades by retransmission, but allow faster communication link during “good” channel states. On the other hand, non-linear slope of SINR curves does not allow for significant rate gain as target error rate moves up to 10%.

	Target PER	**1%**	**5%**	**10%**
DL	Avg. Peak Data Rate [Mbps]	5.7	6.3	6.3
	Peak Data Rate (1-Error Rate)	Ref	+6%	+0.5%
	Effective Data Rate [Mbps]	3.7	4.0	4.0
UL	Avg. Peak Data Rate [Mbps]	2.8	3.1	3.1
	Peak Data Rate (1-Error Rate)	Ref	+6%	+0.7%
	Effective Data Rate [Mbps]	1.9	2.1	2.1
	Coverage	86%	90%	92%

Table 4: Operating error rate vs throughput and coverage.

Furthermore, net throughput, defined as achievable post- channel data rate does not monotonically increase. Compared to the reference error rate of 1%, there is a 6% gain in throughput for 5% PER and only 0.5% gain with an error rate of 10%. Higher packet error rate translates into improved coverage since more users with poorer channel conditions are accepted into system.

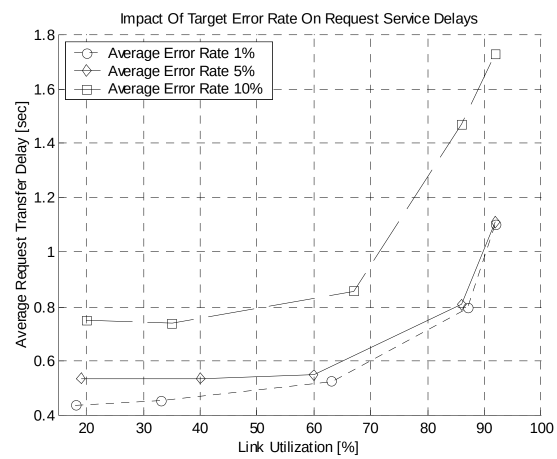

Figure 12. Link layer design permits operating error rate as high as 5%

Request service delays closely relate to wireless link quality and delay bounds determine system operating error rate. The average request service delay as a function of link utilization is plotted in Figure 12 for the three average error rates of interest (listed in Table 3). As expected, 10% ADU error rate and extended fades do not allow ARQ to recover quickly. Other two cases have very close performance, with PER 1% being marginally better across the whole utilization range. From the figure we also read that the system can be safely loaded up to 60% of available bandwidth without any cost paid in service quality. Also, request service delays do not exhibit a sharp knee at any utilization point showing that the system is stable for the whole loading range.

From the above we conclude that TCP throughput and latency requirements for mainstream web applications can easily be obtained with up to 5% ADU loss over the air, due to ARQ/F mechanism. We put the emphasis on *ADU* error rate since without ARQ/F mechanism, unfragmented higher layer packets would experience unacceptable loss rates, forcing TCP to decrease its window often and retransmit excessively. Furthermore, ARQ/F and high allowable error rates permit high spectral efficiency with acceptable delays.

Next, we investigate the effect of Doppler spread on request service delays. **Doppler spread** governs how rapidly channel changes over time and for how long the wireless link stays in a fade. The system with average operating 5% error rate and fairly high link utilization of 65% is examined over two channels with Doppler spreads most typical for fixed wireless access system- low 0.1Hz and high 1Hz. Cumulative distribution function of request service delays is plotted in Figure 13 for the two cases. Request service delays increase in the upper 30% of cases. This is explained by prolonged fade duration for lower Doppler.

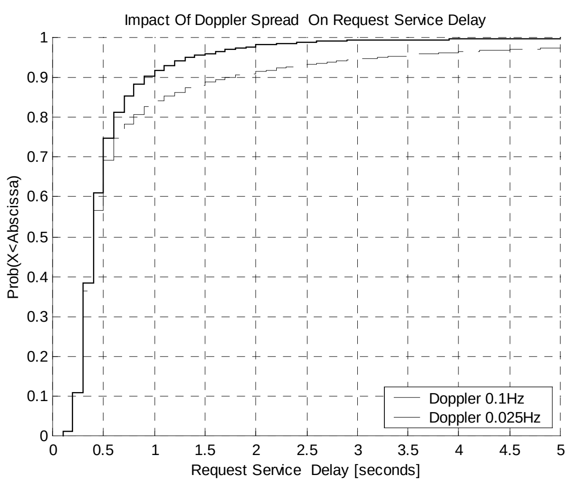

Figure 13. Low Doppler spread and increased fade duration effect around 30% of request downloads

TCP interacts with MAC scheduling and ARQ in several interesting ways. Here we examine the TCP reaction on very slow Doppler with fades up to several seconds. Figure 14 presents sequence number evolution as a function of time and wireless channel. We observe that TCP flow control mechanism restricts the sender during both downlink and uplink fades, since ACKs do not get through either because packet or ACK itself was lost. This effect further shapes applied traffic model. TCP avoids data transmission during fades and more data is attempted when channel improves, as shown on Figure 6. TCP window evolution is slower during burst errors. In a channel that otherwise results in 5% ADU error rate, because of TCP, the observed error rate is below 5%. Random errors confuse TCP, extended errors are correctly identified as network congested or poor channel condition. Furthermore, TCP window evolution slow down during long channel fade prevents MAC buffer overflows. It is seen that link layer ARQ recovers from channel errors efficiently and prevents excessive TCP timeouts, that is, link layer ARQ permits a TCP-friendly design!

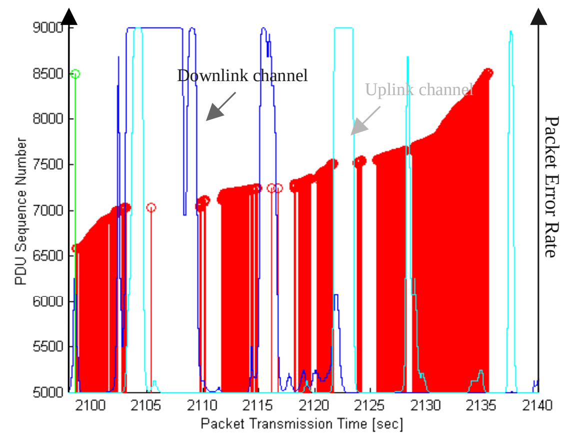

Figure 14. TCP sequence number evolution in fading – Doppler 0.025Hz channel

6. Advanced Features

Other concepts that enhance spectral efficiency of 2G-BWA systems that were not part of the simulation effort presented here, include interference mitigation either at the PHY or MAC, aggressive frequency reuse to ease deployment strategies and channel dependent scheduling to enhance system capacity.

Interference cancellation can be employed on both links to reduce modem setpoints. Array processing techniques at PHY or intelligent scheduling that coordinates transmissions across interfering BTSs lead to a system with aggressive frequency reuse. Additional capacity gain can be achieved with multi-user channel dependent scheduling that allocates bandwidth to the user with the best channel quality while supporting desired QoS.

7. Conclusion

Several advanced technologies that allow broadband wireless service deployment have been described and their benefits have been evaluated. Results show that wireless link shaping has a significant impact on QoS provisioning, and therefore careful design of PHY and MAC layers is critical for acceptable network performance. Adaptive modulation (AM) allows aggressive system design that takes the most advantage of each link condition. While coverage remains guaranteed, AM greatly improves user perceived service quality, measured through average request delay, as well as number of users the system can support. Also, underlying technology that enables finer granularity of AM is multiantenna signal processing at both ends. Our results indicate that MIMO and SD offer additional gain in coverage, almost 2.5 times improvement over SISO system in attainable cell size and data rates, which further benefits QoS. Additionally, the design of link layer, ARQ/F mechanism and MAC scheduling in particular, allow network layer to perform well over wireless channel with packet loss rates as high as 5%. ARQ/F mechanism hides wireless link losses from TCP windowing mechanism, preventing TCP control window restriction. Slower Doppler channels, due to prolonged channel fades, make harder for ARQ/F to quickly recover IP packets, giving rise to TCP timeouts and degrading tail case performance. But, due to acknowledgement based control mechanism, TCP shapes offered traffic load according to wireless channel state, allowing link layer to prevent excessive TCP timeouts and more successfully recover from channel impairments.

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