Project	**IEEE 802.20 Working Group on Mobile Broadband Wireless Access** **<http://grouper.ieee.org/groups/802/20>**
Title	**802.20 Evaluation Methodology Strawman - 00**
Date Submitted	**2003-05-05**
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Re:	MBWA ECSG Call for Contributions
Abstract	This contribution is an attempt to establish a framework for and some initial technical concepts for technology evaluation in the 802.20 project. It continues the process initiated by C802.20-03-24 as presented at the March Meeting of 802.20
Purpose	The intent of this contribution is to establish a working document that will become the repository for the technology evaluation methodology to be used in the selection process for a Draft Standard for 802.20.
Notice	
Release	
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1 INTRODUCTION

As 802.20 embarks upon the important task of developing a new wireless air interface that provides a level of performance far beyond that provided by existing standards that IEEE 802 has developed, 802.20 needs to examine methodologies that are being used in the wireless industry to evaluate proposed technologies. The objective of this contribution is to explain a set of definitions, assumptions and a general framework for simulation proposals submitted for 802.20. With these definitions and assumptions in mind along with the details provided by the owner of any proposal submitted for consideration to 802.20, other people can evaluate such proposal. The scope of the details that must be provided along with each proposal must be such that each proposal can be evaluated ***independently*** by other people.

Over the past few years, system level simulations have emerged as the method of preference to evaluate systems in standards bodies [1]. A number of years ago, link level methods were the prime method. These looked at evaluating the E_b/N_t required on an access terminal to access point basis. While these link level methods are important, it was found that these methods do not adequately capture the richness and dynamics of systems with multiple access points and access terminals. This is particularly true for systems that strive for high capacity, have dynamic control of access terminals, have shared usage of resources, and support data traffic in the multitude of varieties found on the Internet.

What has evolved is a combination of ***link level*** and ***system level*** methodologies. The system level simulation is the main simulation. In the system level simulation, a layout of access points and their antennas are modeled. Access terminals are randomly dropped throughout the defined coverage region. The propagation paths, shadowing, and other blockage between each of the access terminals and the access points are modeled. The fading is updated at some time interval that captures the fading and the shortest control process in the system. For example, the update interval for simulations of WCDMA and cdma2000 are at the power control period (0.67 ms for WCDMA and 1.25 ms for cdma2000). Every update interval, the received signal and noise at each access point and access terminal is determined and the received E_b/N_t is computed. From the received E_b/N_t, some error measures are determined using link level curves. These link level curves are typically generated beforehand so that the execution time of the system level simulation is low.

A system simulation, in general, would be conducted in two phases. The ***first phase*** would cover different mixture of services i.e. voice only, data only, and data plus voice. It would also include physical layer H_ARQ and signaling errors. In general, a configurable fixed number of voice users should be maintain in each simulation run and the data throughput in each cell/sector is evaluated as a function of the number of voice users supported. The data users in each cell/sector shall be assigned one of different traffic models: WAP, HTTP, FTP, and real time video. The ***second phase*** includes TCP and upper layers for data services so that the interaction with TCP can be evaluated.

A typical set of parameters required for a system level simulation for a forward link includes:

1. Number of Cells (3 sectors)
2. Antenna Horizontal pattern, antenna orientation
3. Propagation Model: multi-path channel model, path loss model, shadowing model.
4. Base station correlation
5. Terminal noise figure
6. Thermal noise density
7. Carrier Frequency
8. Base station and terminal antenna gains
9. Other losses
10. Active set parameters
11. Forward link power control
12. Base station maximum Tx power
13. Distance between sites
14. Maximum C/I achievable.

For reverse link system level simulation will have a similar set of parameters. In the next two section we briefly describe different model for propagation environment and traffic models that should be considered in any system level simulation.

2 PROPAGATION ENVIRONMENTS AND FADING MODELS

A number of different propagation models and velocities need to be taken into consideration in both the link level and system level simulations. Previous experience has shown that some aspects of a system design give better performance in some propagation environments and other aspects gives better performance in other environments. Several different channel models are typically considered so that the performance of the system is understood over the set of expected propagation conditions. Furthermore, it is important to avoid system designs that give poor performance in certain environments (e.g., one path Rayleigh fading).

Any particular channel model must specify the number of paths, path delays and power profile, Doppler frequencies, Doppler spectrum for each path, and assignment probability. The channel models are randomly assigned to different users at the beginning of each call and are not changed for the duration of the call. These assignment probabilities represent the percentage of users in each sector with the corresponding channel model. Table 2-1 shows the RMS delay spread and assignment probabilities for each of the environments that we will discuss below.

Table 2-1 Parameters for Channel Impulse Response Models

Enviornment	RMS A (ns)	P(A) %	RMS B (ns)	P(B) %
Indoor	35	50	100	45
Outdoor to Indoor and Pesterian	45	40	750	55
Vehicular	370	40	4000	55

Table 2-2 Indoor Environment Multi-path Channel Model

Tap	Channel A		Channel B		Doppler Spectrum
	Relative Delay (nsec)	Average Power (dB)	Relative Delay (nsec)	Average Power (dB)	
1	0	0	0	0	Flat
2	50	-3.0	100	-3.6	Flat
3	110	-10.0	200	-7.2	Flat
4	170	-18.0	300	-10.8	Flat
5	290	-26.0	400	-18.0	Flat
6	310	-32.0	700	-25.2	Flat

2.1 Indoor Environment

Table 2-2 shows the multi-path channel model for an indoor environment. This environment is characterized by small cells and low transmit powers. Both base stations and pedestrian users are located indoors. The path loss varies due to scattering and attenuation by walls, floors, and metallic structures such as partitions and filing cabinets as well as the metal frame of the building. These objects will also induce shadowing effects. The fading process could range from a pure Rayleigh to Rician, with Doppler frequencies set by walking speeds.

For the indoor environment, the path loss follows the following simplified model

$$L = 37 + 30\log_{10}(R) + 18.3n^{((n+2)/(n+1)-0.46)}$$

where

R is the distance between Tx and Rx in meters

n is the number of floors in the propagation path

L shall be no less than the free space loss. A log-normal shadow fading with a standard deviation of up to 12 dB should be expected. From an interference point of view, this model gives the worst case scenario.

2.2 Outdoor to Indoor and Pedestrian Environment

Table 2-3 shows the multi-path channel model for the Outdoor to Indoor and Pedestrian environment. This environment is characterized by small cells and low transmit power. Base stations with low antenna heights are located outdoors; with a mixture of indoor and outdoor users. Building penetration losses will average 12 dB with a standard deviation of about 8 dB. Rayleigh and/or Rician fading rates will be generally set at walking speeds with occasional faster variations due to moving vehicles.

Table 2-3 Outdoor to Indoor & Pedestrian Environment Multi-path Channel Model

Tap	Channel A		Channel B		Doppler Spectrum
	Relative Delay (nsec)	Average Power (dB)	Relative Delay (nsec)	Average Power (dB)	
1	0	0	0	0	Jakes
2	110	-9.7	200	-0.9	Jakes
3	190	-19.2	800	-4.9	Jakes
4	410	-22.8	1200	-8.0	Jakes
5	-	-	2300	-7.8	Jakes
6	-	-	3700	-23.9	Jakes

A propagation path loss proportional to R^{-4} should be expected, but a wide range of path loss exponents should be considered. For example, if the path is a line of sight (LOS) path on a canyon-like street, for example, the path loss will follow R^{-2} where there is Fresnel zone clearance. For regions where there is no Fresnel zone clearance a path loss proportional to R^{-4} should be expected but it could be also range up to R^{-6} due to trees and other obstructions along the propagation path. In general, the following model shall be used for the propagation path loss

$$L = 40\log_{10}(R) + 30\log_{10}(f) + 49$$

where

R is the distance between the Tx and Rx

f is the carrier frequency in MHz

L shall be no less than the free space loss. This model is valid for non-LOS propagation only and describes a worst case propagation and hence should be used for coverage efficiency evaluation purposes only. Log-normal shadowing with a standard deviation of 10 dB for outdoors and 12 dB for indoors should be expected.

The model described below is a more detailed model that considers both LOS and non-LOS propagation. This model assumes a Manhattan-like, i.e. urban, environment, and hence, it could be used for spectral efficiency evaluation. This model is a recursive model that calculates the path loss as a sum of LOS and NLOS segments. The shortest path along streets between the base station and the terminal is found within the Manhattan environment. In this case the path loss is given by

$$L = 20 \cdot \log_{10} \frac{4\pi d_n}{\lambda}$$

where d_n is the "effective distance" and λ is the wavelength, n is the number of street segments between BS and MS along the shortest path (see [2] for the details of calculating d_n). This model can be extended to cover micro cell dual slope behavior, by modifying the expression to

$$L = 20 \cdot \log_{10} \left(\frac{4\pi d_n}{\lambda} \right)^{D(x)} \qquad x = \sum_{j=1}^{n} s_{j-1}$$

$$D(x) = \begin{cases} x / x_b & x > x_b \\ 1 & x \le x_b \end{cases}$$

where s_j is the length in meters of the j-th segment and x_b is the break point which is set to 300 meters. So the slope before the break point is 2 and increases to 4 after the break point. Moreover, to account for the effects of propagation above roof tops (shortest geographical distance), we use the Walfish-Ikegami Model with antenna below roof tops

$$L = 24 + 45 \log_{10}(d + 20)$$

where d is the shortest physical distance between the transmitter and receiver in meters. The final path loss is the minimum of the "Manhatten path loss" and the path loss based on the shortest path loss.

2.3 Vehicular Environment

Table 2-4 shows the multi-path channel model for a vehicular environment. Here, the environment is characterized by larger cells and high transmit power. Rayleigh fading rates are set by the vehicle speed with occasional lower rates for stationary terminals. A geometrical path loss proportional to R^{-4} and a log-normal shadowing with a standard deviation of 10 dB is suitable for urban and sub-urban environments. In flat terrains the path loss is lower than that of urban and sub-urban environments. In general, the path loss for the vehicular environment is

$$L = 40\left(1 - 0.004 \times \Delta h_b\right) \log_{10}(R) - 18 \log_{10}\left(\Delta h_b\right) + 21 \log_{10}(f) + 80 \text{ dB}$$

where

R is the distance between the transmitter and receiver in kilometers and f is the carrier frequency in MHz, and Δh_b is the base station antenna height in meters from the average roof top level which is assumed to be between 0 and 50 meters. For a Δh_b of 15 meters and a 2GHz frequency, the pass loss reduces to

$$L = 128.1 + 37.6\log_{10}(R)$$

Again, L shall be less than the free space loss. This model is valid for NLOS propagation only and represents a worst case propagation scenario.

Table 2-4 Vehicular Environment Multi-path Channel Model

Tap	Channel A		Channel B		Doppler Spectrum
	Relative Delay (nsec)	Average Power (dB)	Relative Delay (nsec)	Average Power (dB)	
1	0	0	0	-2.5	Jakes
2	310	-1.0	300	0	Jakes
3	710	-9.0	8900	-12.8	Jakes
4	1090	-10.0	12900	-10.0	Jakes
5	1730	-15.0	17100	-25.2	Jakes
6	2510	-20.0	20000	-16.0	Jakes

2.4 Shadow Fading Temporal Decorrelation

The shadow fading around the mean path loss L in dB is characterized by a Gaussian distribution with zero mean and a standard deviation that depends on the propagation environment. The shadowing is also a slow fading process with distance Δx (or equivalently time) and hence adjacent fading values are correlated. The autocorrelation function of the log term fading can be described by

$$\rho(\Delta x) = e^{-0.69315.|\Delta x|/x_c}$$

where x_c is the decorrelation distance which is different for each environment. This model is more accurate for sub-urban environment.

2.5 Antenna Patterns for Sectorization

A key component in modeling the propagation channel between the Tx and Rx is the antenna pattern used for sectorization. A suitable model for this pattern is given by [1]

$$A(\theta) = -\min\left\{12\left(\frac{\theta}{\theta_{3dB}}\right)^2, A_m\right\} \qquad -180 \le \theta \le 180$$

where θ_{3dB} is the 3 dB beamwidth and A_m =20 dB is the maximum attenuation.

3 TRAFFIC MODELS

In the past, when systems were primarily evaluated at the link level, the traffic model was not considered. This was equivalent to what is called in system level modeling, "full buffer" analysis, which assumes that there is an infinite source of data between each access terminal and access point. While an important component of analysis, the "full buffer" model neglects many important aspects of a system design. Neglecting these aspects has lead to designs that do not perform well when subjected to the variety of traffic found in commercial systems. Thus, it is important to model any candidate system using real traffic models. Some that should be used include FTP, web browsing (HTTP), OMA (WAP), video, and voice. Note that even for these services, the access terminal to access point traffic is different than the access point to access terminal (e.g., for web browsing, most of the access terminal to access point traffic is http requests and the access point to access terminal traffic is web page downloads). It is also important to include the effects of TCP: improper system design can lead to poor performance, not as a result of problems in the link but due to unintended interactions with TCP.

As we mentioned earlier, a data user in any given sector can have one of 4 different traffic models. So, in any given sector, there will be a mixture of different data users, each having a different traffic model. Hence, it is also necessary to have a probability of assignment for each data traffic model which would represent the percentage number of users that will be using that particular data traffic model with a cell/sector. A typical values for these assignment probabilities is shown in Table 3-1.

Table 3-1 Assignment Probability for Different Traffic Models

HTTP	FTP	WAP	Real Time Video
24.43%	9.29%	56.43%	9.85%

3.1 TCP Model

Both HTTP and FTP use TCP as their transport protocol, we briefly describe a model for TCP traffic that will serve as base for both the HTTP and FTP traffic models described later.

The TCP connection set-up and release protocols use a three-way handshake mechanism as described in Figure 1. The amount of outstanding data that can be sent without receiving an acknowledgement (ACK) is determined by the minimum of the congestion window size and the receiver window size. After the connection setup, the transfer of data starts in slow-start mode with an initial congestion window size of 1 segment. The congestion window is increased by one segment for each ACK packets received by the sender regardless of whether the packet is correctly received or not, and regardless of whether the packet is out of order or not. This results in an exponential growth of the congestion window.

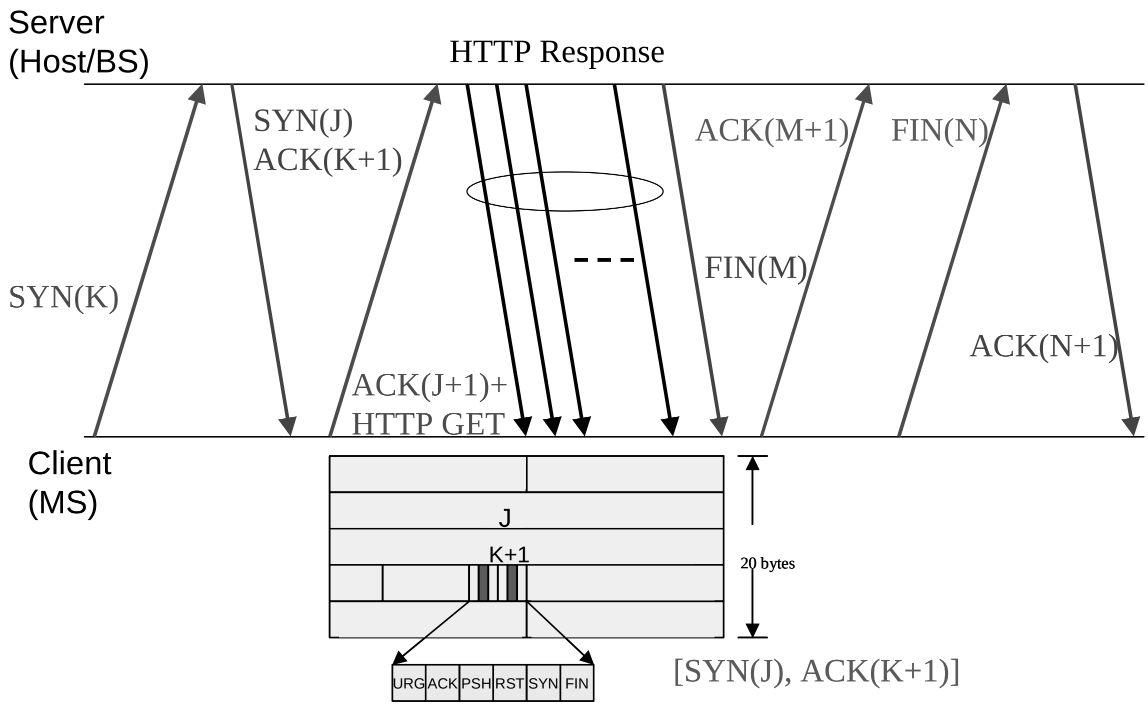

Figure 1 Control in TCP Connection Setup and Release

Let τ_{rt} denote the round trip time. This round trip time consists of two parts, that is

$$\tau_{rt} = \tau_c + \tau_l$$

where τ_c is the sum of time taken by an ACK packet to travel from the client to the server and the time taken by a TCP data segment to travel from the server to the base station router, and τ_l is the transmission time taken for a TCP data segment to travel from the base station router to the client. In other words, the round trip time can be seen as the total time between sending an ACK packet from the client and receiving the first packet transmitted in response of the ACK packet. In this model, τ_c is modeled as an exponentially distributed random variable (the mean of this distribution μ is a simulation parameter) and τ_l is determined by the available access link throughput. For the purpose of the ***first phase*** of system simulation, we may ignore the detailed specifics of the TCP congestion control and avoidance. In addition, the receive window size (RSWIN) can be assumed to be large enough and hence will not be a limitation. Thus at the base station, after every successful packet transmission, two data segments arrive at the base back-to-back.

3.2 FTP Traffic Models

In FTP applications, a session consists of a sequence of file transfers, separated by *reading times*. The two main parameters of an FTP session are:

1. S : the size of a file to be transferred
2. D_{pc} : reading time, i.e., the time interval between end of download of the previous file and the user request for the next file.

The underlying transport protocol for FTP is TCP. The packet trace of an FTP session is shown in
Figure **2**. The parameters for the FTP application session are described in Table 3-2.

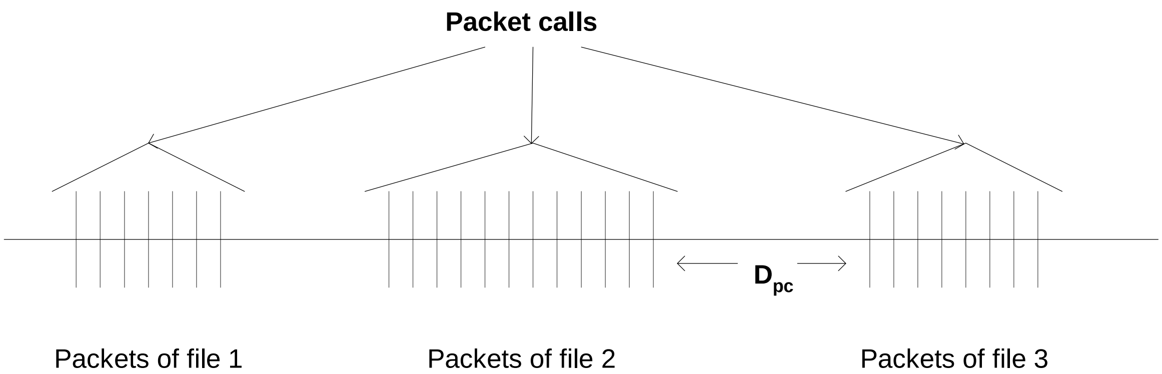

Figure 2 Packet Trace in a Typical FTP Session

Table 3-2 FTP Traffic Model Parameters

Component	**Distribution**	**Parameters**	**PDF**
File size (S)	Truncated Lognormal	Mean = 2Mbytes Std. Dev. = 0.722 Mbytes Maximum = 5 Mbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x}\exp\left[\frac{-(\ln x-\mu)^2}{2\sigma^2}\right], x \geq 0$ $\sigma = 0.35, \mu = 14.45$
Reading time (D_{pc})	Exponential	Mean = 180 sec.	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.006$

Based on the results on packet size distribution 76% of the files are transferred using and MTU of 1500 bytes and 24% of the files are transferred using an MTU of 576 bytes. For each file transfer a new TCP connection is used whose initial congestion window size is 1 segment (i.e. MTU). The packet arrival process at the base station is described by the TCP model described earlier. The process for generation of FTP traffic is described Figure 3.

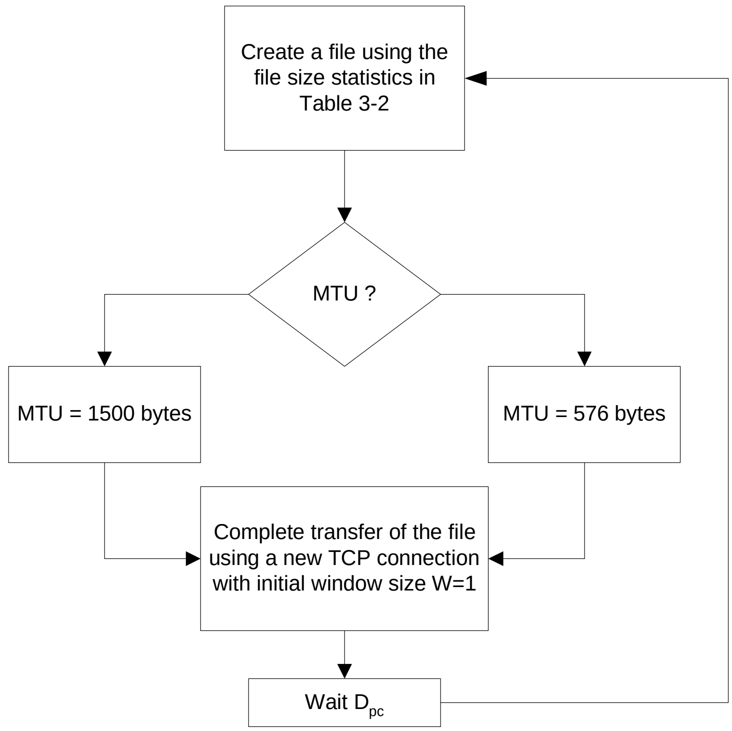

Figure 3 Model for generatinig FTP traffic

3.3 HTTP Traffic Model

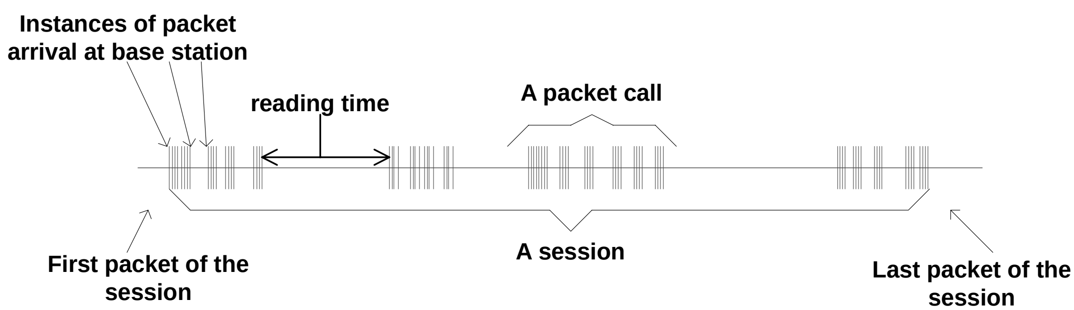

Figure 4 Packet Trace of a Typical Web Browsing Session

Figure 4 shows the packet trace of a typical web browsing session. The session is divided into ON/OFF periods representing web-page downloads and the intermediate reading times. In Figure 4, the web-page downloads are referred to as packet calls. These

ON and OFF periods are a result of human interaction where the packet call represents a user's request for information and the reading time identifies the time required to digest the web-page.

As is well known, web-browsing traffic is self-similar. In other words, the traffic exhibits similar statistics on different timescales. Therefore, a packet call, like a packet session, is divided into ON/OFF periods as in Figure 5**. Unlike a packet session, the ON/OFF periods within a packet call are attributed to machine interaction rather than human interaction. As an example, consider a typical web-page from the Wall Street Journal (WSJ) Interactive edition depicted in Figure 6. This web-page is constructed from many individually referenced objects. A web-browser will begin serving a user's request by fetching the initial HTML page using an HTTP GET request. After receiving the page, the web-browser will parse the HTML page for additional references to embedded image files such as the graphics on the tops and sides of the page as well as the stylized buttons. The retrieval of the initial page and each of the constituent *objects* is represented by ON period within the packet call while the parsing time and protocol overhead are represented by the OFF periods within a packet call. For simplicity, the term "page" will be used in this paper to refer to each packet call ON period. As a rule-of-thumb, a page represents an individual HTTP request explicitly initiated by the user. The initial HTML page is referred to as the "main object" and the each of the constituent objects referenced from the main object are referred to as an "embedded object".**

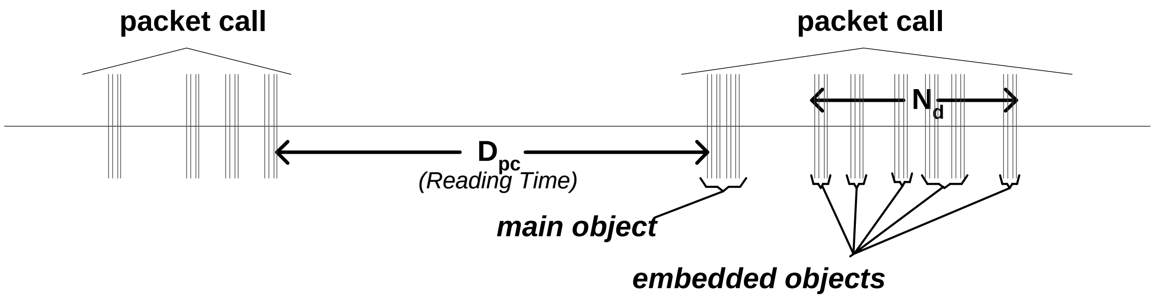

Figure 5 Contents in a Packet Call

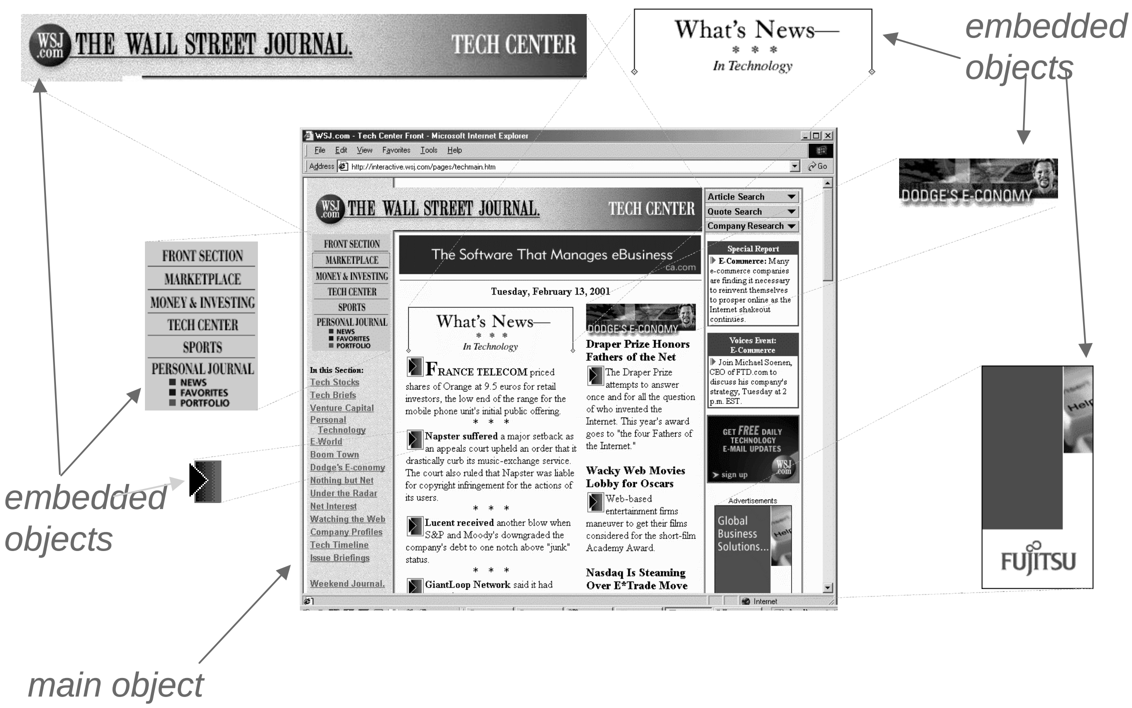

Figure 6 A Typical Web Page and its Content

The parameters for the web browsing traffic are as follows:

- **S_M: Size of the main object in a page**
- **S_E: Size of an embedded object in a page**
- **N_d: Number of embedded objects in a page**
- **D_{pc}: Reading time**
- **T_p: Parsing time for the main page**

The packet traffic characteristics within a packet call will depend on the version of HTTP used by the web servers and browsers. Currently two versions of the protocol, HTTP/1.0 and HTTP/1.1, are widely used by the servers and browsers. These two versions differ in how the transport layer TCP connections are used for the transfer of the main and the

embedded objects as described below and ***hence each will have its own packet arrival model.***
In HTTP/1.0, a distinct TCP connection is used for each of the main and embedded objects downloaded in a web page. Most of the popular browser clients download the embedded objects using multiple simultaneous TCP connections; this is known as *HTTP/1.0-burst mode transfer*. The maximum number of such simultaneous TCP connections, N, is configurable; most browsers use a maximum of 4 simultaneous TCP connections. If there are more than N embedded objects, a new TCP connection is initiated when an existing connection is closed. The effects of slow-start and congestion control overhead of TCP occur on a per object basis.
In HTTP/1.1, persistent TCP connections are used to download the objects, which are located at the same server and the objects are transferred serially over a single TCP connection; this is known as *HTTP/1.1-persistent mode transfer*. The TCP overhead of slow-start and congestion control occur only once per persistent connection.

The distributions of the parameters for the web browsing traffic model are described in

Table 3-3 HTTP Traffic Model Parameters

Component	Distribution	Parameters	PDF
Main object size (S_M)	Truncated Lognormal	Mean = 10710 bytes Std. dev. = 25032 bytes Minimum = 100 bytes Maximum = 2 Mbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x - \mu)^2}{2\sigma^2}\right], x \geq 0$ $\sigma = 1.37, \mu = 8.35$
Embedded object size (S_E)	Truncated Lognormal	Mean = 7758 bytes Std. dev. = 126168 bytes Minimum = 50 bytes Maximum = 2 Mbytes	$f_x = \frac{1}{\sqrt{2\pi}\sigma x} \exp\left[\frac{-(\ln x - \mu)^2}{2\sigma^2}\right], x \geq 0$ $\sigma = 2.36, \mu = 6.17$
Number of embedded objects per page (N_d)	Truncated Pareto	Mean = 5.64 Max. = 53	Note: Subtract k from the generated r.v. to get N_d $f_x = \frac{\alpha k^\alpha}{x^{\alpha+1}}, k \leq x < m$ $f_x = \left(\frac{k}{m}\right)^\alpha, x = m$ $\alpha = 1.1, k = 2, m = 55$
Reading time (D_{pc})	Exponential	Mean = 30 sec	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 0.033$
Parsing time (T_p)	Exponential	Mean = 0.13 sec	$f_x = \lambda e^{-\lambda x}, x \geq 0$ $\lambda = 7.69$

Note: When generating a random sample from a truncated distribution, discard the random sample when it is outside the valid interval and regenerate another random sample.

3.4 WAP Traffic Model

Each WAP request from the browser is modeled as having a fixed size and causes the WAP server to send back a response with an exponentially distributed response time. The WAP gateway response time is the time between when the last octet of the request is sent and when the first octet of the response is received from the WAP server. The response itself is composed of a geometrically distributed number of objects, and the inter-arrival time between these objects is exponentially distributed. Once the last object is received, the exponentially distributed reading time starts, and it ends when the WAP browser generates the next request. Table 3-4 describes the distribution of the model parameters. During the simulation period, the model assumes that each WAP user is continuously active, i.e., making WAP requests, waiting for the response, waiting the reading time, and then making the next request.

Table 3-4 WAP Traffic Model Parameters

Packet based information types	Size of WAP request	Object size	# of objects per response	Inter-arrival time between objects	WAP gateway response time	Reading time
Distribution	Deterministic	Truncated Pareto (Mean= 256 bytes, Max= 1400 bytes)	Geometric plus offset of 1	Exponential	Exponential	Exponential
Distribution Parameters	76 octets	K = 71.7 bytes, $\alpha = 1.1$	Mean = 2 plus offset of 1	Mean = 1.6 s	Mean = 2.5 s	Mean = 5.5 s

3.5 Near Real-Time Video Traffic Model

The following section describes a model for streaming video traffic on the forward link. Figure 7 describes the steady state of video streaming traffic from the network as seen by the base station. Latency of starting up the call is not considered in this steady state model.

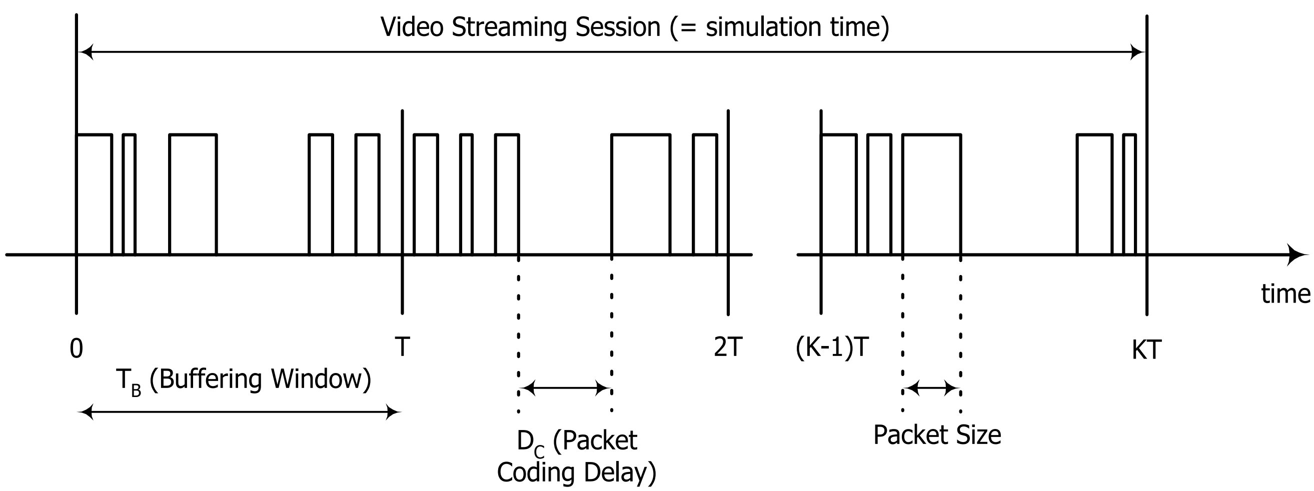

Figure 7 Near Real-Time Video Traffic Model

A video streaming session is defined as the entire video streaming call time, which is equal to the simulation time for this model.

Each frame of video data arrives at a regular interval T determined by the number of frames per second (fps). Each frame is decomposed into a fixed number of slices, each

transmitted as a single packet. The size of these packets/slices is distributed as a truncated Pareto. Encoding delay, Dc, at the video encoder introduces delay intervals between the packets of a frame. These intervals are modeled by a truncated Pareto distribution. The parameter T_B is the length (in seconds) of the de-jitter buffer window in the mobile station used to guarantee a continuous display of video streaming data. This parameter is not relevant for generating the traffic distribution but is useful for identifying periods when the real-time constraint of this service is not met. At the beginning of the simulation, it is assumed that the mobile station de-jitter buffer is full with (T_B x source video data rate) bits of data. Over the simulation time, data is "leaked" out of this buffer at the source video data rate and "filled" as forward link traffic reaches the mobile station. As a performance criterion, the mobile station can record the length of time, if any, during which the de-jitter buffer runs dry. The de-jitter buffer window for the video streaming service is 5 seconds.
Using a source video rate of 32 kbps, the video traffic model parameters are defined Table 3-5

Table 3-5 Near Real-Time Video Traffic Model Parameters

Information types	Inter-arrival time between the beginning of each frame	Number of packets (slices) in a frame	Packet (slice) size	Inter-arrival time between packets (slices) in a frame
Distribution	Deterministic (Based on 10fps)	Deterministic	Truncated Pareto (Mean= 50bytes, Max= 125bytes)	Truncated Pareto (Mean= 6ms, Max= 12.5ms)
Distribution Parameters	100ms	8	K = 20bytes α = 1.2	K = 2.5ms α = 1.2

3.6 Voice Traffic Model

The voice traffic model will depend on the voice codec used as well as whether voice in 802.20 will be implemented as a circuit switched or voice over IP. Voice will in general follow a Markov source model with different rates (full rate, half rate, etc) with a corresponding set of transition probabilities between different rates. Voice capacity is obtained based on satisfying a certain outage criteria (or a group of), for example short term FER, per user outage, and/or system outage.

4 SUMMARY

802.20 should put together an analysis framework for evaluating the systems designs that it develops. This analysis framework should include system level simulations with realistic propagation models and traffic models.

5 REFERENCES

[1] 3GPP2 C.R.1002, *1xEV-DV Evaluation Methodology (v10).*

[2] ETRI ETR 112, *Selection Procedure for the Choice of Radio Transmission Technologies for UMTS*