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1 Traffic Models

Applications

Web Browsing (HTTP) Traffic Model

HTTP traffic characteristics are governed by the structure of the web pages on the World Wide Web (WWW), and the nature of human interaction. The nature of human interaction with the WWW causes the HTTP traffic to have a bursty profile, where the HTTP traffic is characterized by ON/OFF periods as shown in Figure 1-1.

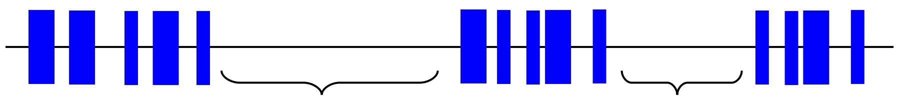

Figure 1-1: HTTP Traffic Pattern

The ON periods represent the sequence of packets in which the web page is being transferred from source to destination; while the OFF periods represent the time the user spends reading the webpage before transitioning to another page. This time is also known as Reading Time ,.

The amount of information passed from the source to destination during the ON period is governed by the web page structure. A webpage is usually composed of a main object and several embedded objects. The size of the main object, in addition to the number and size of the embedded objects define the amount of traffic passed from source to destination.

In summary, the HTTP traffic model is defined by the following parameters:

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

In addition to the model parameters, HTTP traffic behavior is also dependent on the HTTP version used. Currently HTTP 1.0 and HTTP 1.1 are widely used by servers and browsers [, . In HTTP 1.0, also known as burst mode transfer, a distinct TCP connection is used for each object in the page, thereby facilitating simultaneous transfer of objects. The maximum number of simultaneous TCP connections is configurable, with most browsers using a maximum of 4 simultaneous TCP connections. In HTTP/1.1, also known as persistent mode transfer, all objects are transferred serially over a single persistent TCP connection. Table 1-1 provides the model parameters for HTTP traffic for downlink and uplink connections , .

Component	Distribution	Parameters Downlink	 Uplink	PDF
Main object size (S_M)	Truncated Lognormal	Mean = 10710 bytes SD= 25032 bytes Min = 100 bytes Max = 2 Mbytes 1.37, 8.35	Mean = 9055 bytes SD = 13265 bytes Min = 100 bytes Max = 100 Kbytes 1.37, 8.35	$f_x \quad \frac{1}{\sqrt{2} \quad x} \exp \quad \frac{\ln x \qquad ^{2}}{2 \quad ^{2}}, x \quad 0$ if x>max or x<min, discard and generate a new value for x
Embedded object size (S_E)	Truncated Lognormal	Mean = 7758 bytes SD = 126168 bytes Min = 50 bytes Max = 2 Mbytes 2.36, 6.17	Mean = 5958 bytes SD = 11376 bytes Min = 50 bytes Max = 100 Kbytes 1.69, 7.53	$f_x \quad \frac{1}{\sqrt{2} \quad x} \exp \quad \frac{\ln x \qquad ^{2}}{2 \quad ^{2}}, x \quad 0$ if x>max or x<min, discard and generate a new value for x
Number of embedded objects per page (N_d)	Truncated Pareto	Mean = 5.64 Max. = 53	Mean = 4.229 Max. = 53 $1.1, k \quad 2,$ $m \quad 55$	$f_x \quad \frac{k}{x \quad ^{1}}, k \quad x \quad m$ $f_x \quad \frac{k}{m}, x \quad m$ Subtract k from the generated random value to obtain N_d if x>max, discard and regenerate a new value for x
Reading time (D_{pc})	DL: Exponential UL: Uniform	Mean = 30 sec	Mean = 5 sec 0.033 $a \quad 0 \qquad b \quad 10$	$DL: \; f_x \quad e^{\quad x}, x \quad 0 \quad UL: \; f_x \quad \frac{1}{b \quad a}, a \quad x \quad b$
Parsing time (T_p)	Exponential	Mean = 0.13 sec	Mean = 0.13 sec 7.69	$f_x \quad e^{\quad x}, x \quad 0$

Table 1-1: HTTP Traffic Model

To request an HTTP session, the client sends an HTTP request packet, which has a constant size of 350 bytes

From the statistics presented in the literature, a 50%-50% distribution of HTTP versions between HTTP 1.0 and HTTP 1.1 has been found to closely approximate web browsing traffic in the internet .

Further studies also showed that the maximum transmit unit (MTU) sizes most common to in the internet are 576 bytes and 1500 bytes (including the TCP header) with a distribution of 24% and 76% respectively. Thus, the web traffic generation process can be described as in Figure 1-2.

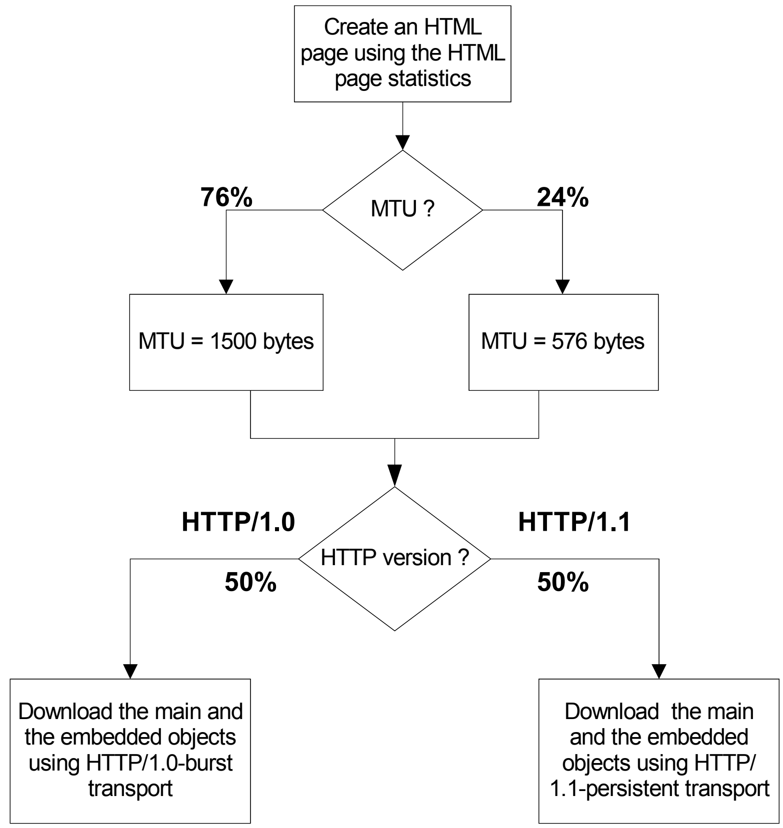

Figure 1-2: HTTP Traffic Profiles

File Transfer (FTP) Traffic Model

File transfer traffic is characterized by a session consisting of a sequence of file transfers, separated reading times. Reading time is defined as the time between end of transfer of the first file and the transfer request for the next file. The packet call size is therefore equivalent to the file size and the packet call inter-arrival time is the reading time. A typical FTP session is shown in Figure 1-3.

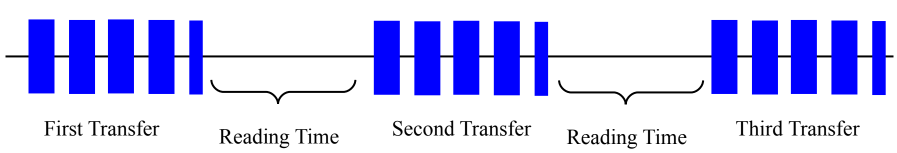

Figure 1-3: FTP Traffic Patterns

Table 1-2 provides the model parameters for FTP traffic that includes file downloads as well as uploads . In the case of file uploads, the arrival of new users is Poisson distributed and each user transfers a single file before leaving the network. The FTP traffic generation process is described in Figure 1-4.

Component	Distribution	Parameters Download	Parameters Upload	PDF
File size (S)	Truncated Lognormal	Mean = 2Mbytes SD = 0.722 Mbytes Max = 5 Mbytes 0.35 14.45	Min = 0.5 Kbytes Max = 500 Kbytes Mean = 19.5 Kbytes SD = 46.7 Kbytes 2.0899 0.9385	$f_x \quad \frac{1}{\sqrt{2} \quad x} \exp \quad \frac{\ln x \qquad ^2}{2 \quad ^2} \quad , x \quad 0$ if x>max or x<min, discard and generate a new value for x
Reading time (D_{pc})	Exponential	Mean = 180 sec. 0.00556	N/A	Download: $f_x \quad e^{\quad x} \quad , x \quad 0$ Upload: N/A

Table 1-2: FTP Traffic Model

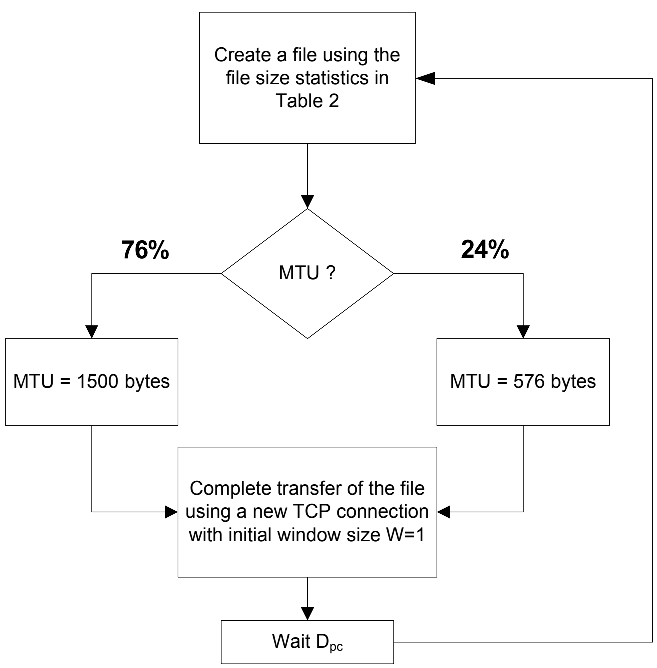

Figure 1-4: FTP Traffic Profiles

Speech Source Model (VoIP)

A VoIP user is in outage (not satisfied) if 97% radio interface tail latency of this user is greater than 10 ms. This assumes an end-to-end delay less than 200 ms for mobile-to-mobile communications. The system capacity is defined as the number of users in the cell when more than 97% of the users are satisfied [32].

Erasure rate for consecutive full rate AMR voice frames shall be less than 3%. The following table provides the relevant parameters of the VoIP traffic that shall be assumed in the simulations. The details of the corresponding traffic model are described below:

Parameter	*Characterization*
Codec	RTP AMR 12.2, Source rate 12.2 kbps
Encoder frame length	20 ms
Voice activity factor (VAF)	~ 40% (c=0.01, d=0.985)
SID payload	Modeled 15 bytes (5 Bytes + header) SID packet every 160 ms during silence
Protocol Overhead with compressed header	10 bit + padding (RTP-pre-header) 4 Byte (RTP/UDP/IP) 2 Byte (RLC/security) 16 bits (CRC)
Total voice payload on air interface	40 bytes (AMR 12.2)

Table 1-3: Detailed description of the VoIP traffic model

1.1.1.1 Basic Voice Model

Consider the simple 2-state voice activity Markov model shown in .

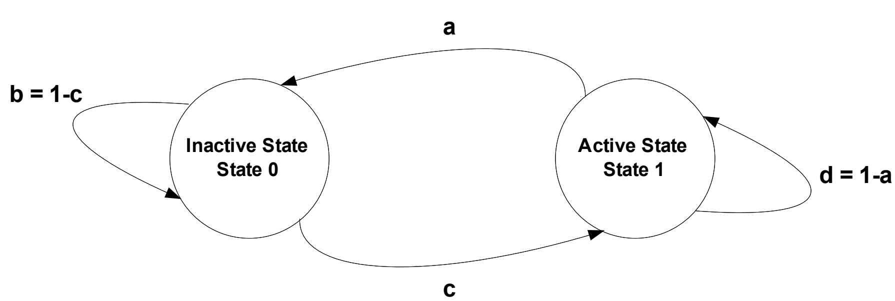

Figure 1-5: 2-state voice activity Markov model (after [32])

In the model, the conditional probability of transitioning from state 1 (the active speech state) to state 0 (the inactive or silent state) while in state 1 is equal to a, while the conditional probability of transitioning from state 0 to state 1 while in state 0 is c. The model is assumed updated at the speech encoder frame rate $R = 1/T$, where T is the encoder frame duration (typically, 20ms).

1.1.1.2 Basic Model Statistics

The steady-state equilibrium of the model requires that

$$P_0 = \frac{a}{a+c}$$
$$P_1 = \frac{c}{a+c} \qquad (1.1\text{-}1)$$

Where P_0 and P_1 are respectively the probability of being in state 0 and state 1. The Voice Activity Factor (VAF) l is given by

$$l = P_1 = \frac{c}{a+c} \qquad 1.1\text{-}2)$$

A talk-spurt is defined as the time period t_{TS} between entering the active state (state 1) and leaving the active state. The probability that a talk spurt has duration n speech frames is given by

$$P_{t_{TS}=n} @ P_{TS}(n) = a(1-a)^{n-1} \quad n = 1,2,\cdots \qquad 1.1\text{-}3)$$

Correspondingly, the probability that a silence period has duration n speech frames is given by

$$P_{t_{SP}=n} @ P_{SP}(n) = c(1-c)^{n-1} \quad n = 1,2,\cdots \qquad 1.1\text{-}4)$$

The mean talk spurt duration m_{TS} (in speech frames) is given by

$$m_{TS} = E[t_{TS}] = \frac{1}{a} \qquad (1.1\text{-}5)$$

While the mean silence period duration m_{SP} (in speech frames) is given by

$$m_{SP} = E[t_{SP}] = \frac{1}{c} \quad 1.1\text{-}6)$$

The distribution of the time period t_{AE} (in speech frames) between successive active state entries is the convolution of the distributions of t_{SP} and t_{TS}. This is given by

$$P_{t_{AE}=n} @ P_{AE}(n) = \frac{c}{c-a} a(1-a)^{n-1} + \frac{a}{a-c} c(1-c)^{n-1} \quad n = 1,2,\cdots \qquad 1.1\text{-}7)$$

Note that t_{AE} can be used as a rough estimate of the time between MAC layer resource reservations, provided a single reservation is made per user per talk-spurt. Note that in practice, very small values of t_{AE} might not lead to a separate reservation request, but equation (1.1-7) still offers some potentially useful guidance.

Since the state transitions from state 1 to state 0 and vice versa is independent, the mean time m_{AE} between active state entries is given simply by the sum of the mean time in each state. That is

$$m_{AE} = m_{TS} + m_{SP} \quad 1.1\text{-}8)$$

Accordingly, the mean rate of arrival $\bar{R}_{AE}$ of transitions into the active state is given by

$$\bar{R}_{AE} = \frac{1}{m_{AE}} \quad 1.1\text{-}9)$$

As a simple example, consider the case where the speech encoder-frame duration $T = 20ms$. Further assume a desired VAF of 60% ($l = 0.6$), and a desired mean talk spurt duration of 5 s. Therefore, from equation (1.1-5), $1/a = 5/T$ and so $a = 0.04$. Further, from equation (1.1-2), $c = al/(1-l) = 0.006$.

For these parameters, the resulting theoretical and simulated distributions of the talk spurt duration (t_{TS}, in seconds), silence period duration (t_{SP}, in seconds), and time between active state entry (t_{AE}, in seconds) appear in Figure 1-6.

The mean talk spurt duration is given by $m_{TS} = 1/a = 250$ frames, or 5 s. Correspondingly, the mean silence period duration is $m_{SP} = 1/c = 166.67$ frames, or 3.33 s. The resulting mean time between active state entry is then 8.33 s, and so the mean rate of arrival of talk spurts is $\bar{R}_{AE} = 1/8.33 = 0.12$ talk spurts per second.

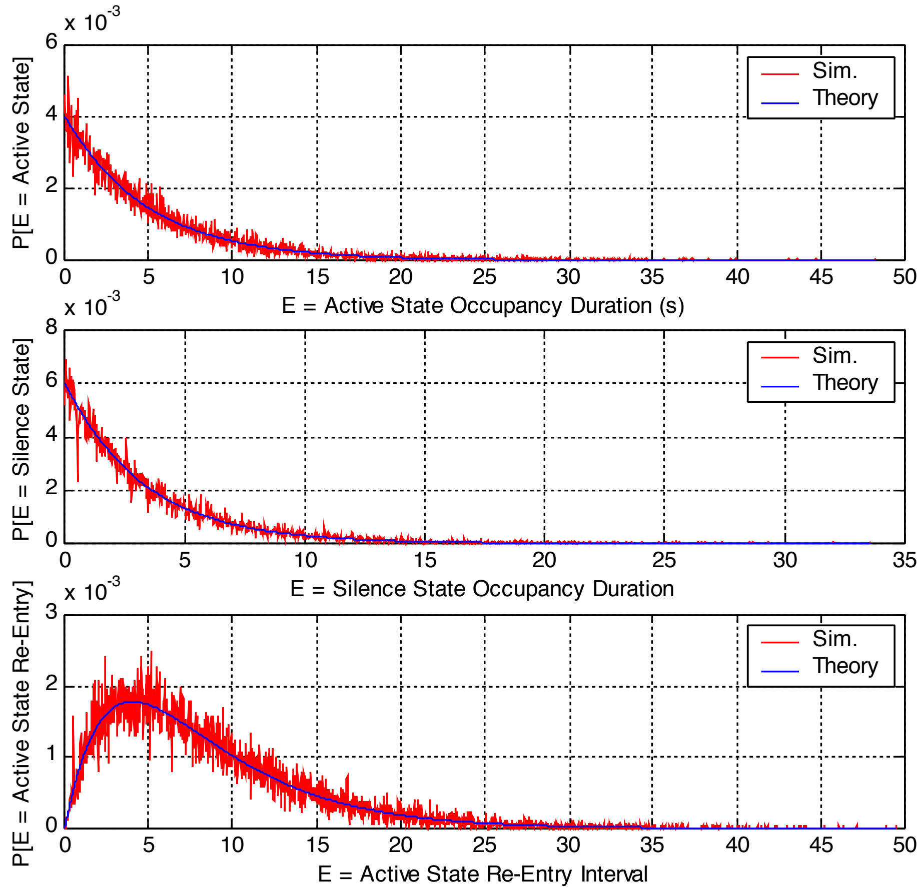

Figure 1-6 – State duration and entry distributions – theory and simulation (after [32]).

The simplified speech source model with an average voice activity of 0.4 is given by

```
IF PrevState=0 then
    IF RAND () <0.01 then
            NewState=1 /* go to voice active state */
    Else
            NewState=0 /* remain in voice inactive state */
Else
    IF RAND () <0.985 then
            NewState=1 /* remain in voice active state */
    Else
            NewState=0 /* go to voice inactive state */
```

Voice users should meet an outage criterion which can be defined as:

a. Average FER being less than 3%,

b. Short-term FER exceeding 3% no more than 3% of the time.

The short-term FER of the voice service is calculated by averaging over 2 seconds. An AMR vocoder with a rate of 12.2 kbps will be used. The uplink voice activity factor should be set to 0.32 by randomly choosing on and off periods of appropriate duration. A simple speech source model is given above.

During the active state, packets of fixed sizes are generated at a regular interval. During the inactive state, we model comfort noise generation with smaller packet sizes at a regular interval instead of no packet transmission. The size of packet and the rate at which the packets are sent depends on the corresponding voice codecs and compression schemes. Table 1-4 provides information on some common vocoder.

Vocoder			EVRC	AMR	GSM 6.10	G.711	G.723.1		G729A
Source Bit rate [Kb/s]			0.8/2/4/8.55	4.75-12.2	13	64	5.3	6.4	8
Frame duration [ms]			20	20	20	10	30	30	10
Information bits per frame			16/40/80/171	95-244	260	640	159	192	80
RTP/ UDP/IP header [bytes]	IPv4	Uncompressed Header	40	40	40	40	40	40	40
		Compressed Header	2	2	2	2	2	2	2
	IPv6	Uncompressed Header	60	60	60	60	60	60	60
		Compressed Header	4	4	4	4	4	4	4

Table 1-4: Vocoder typical rates and packet sizes

To calculate the total packet size, MAC headers and CRC need to be accounted for (example: there are 6 bytes of MAC header and CRC in IEEE 802.16e reference system). For example, without header compression, an AMR payload of 33 bytes is generated in the active state for every 20 ms. SID frames constitute the AMR payload of 7 bytes every 160 ms in the inactive state, resulting in a packet size of 83 (57) bytes for the active (inactive) mode, respectively, assuming IPv4 and uncompressed headers.

Near Real Time Video Streaming

A video streaming session is composed of a series of frames that arrive at a regular interval of T, which is determined by the number of frames per second. Each frame consists of a fixed number of packets whose size is distributed as a truncated Pareto.

The delay between packets is caused by the encoder, and the delay distribution is modeled by a truncated Pareto distribution. To minimize the effects of the non-uniform delay between the packets, a buffer is used at the end point of the session to guarantee a continuous and smooth display of the video streaming data. For video streaming services, this buffer is set to 5 seconds.

Table 1-5 provides the model parameters for video streaming traffic for a video source rate of 32 kbps, and 10 frames per second:

Component	Distribution	Parameter values	PDF
Frame inter-arrival time	Constant	100 ms	Deterministic
Packets per frame	Constant	8	Deterministic
Packet size	Truncated Pareto	Mean = 50 bytes Max. = 125 bytes K = 20 = 1.2	$f_x \quad \frac{k}{x^{\quad 1}}, k \quad x \quad m$ $f_x \quad \frac{k}{m} \quad , x \quad m$
Packet inter-arrival time	Truncated Pareto	Mean = 6 ms Max = 12.5 ms K = 2.5 ms = 1.2	$f_x \quad \frac{k}{x^{\quad 1}}, k \quad x \quad m$ $f_x \quad \frac{k}{m} \quad , x \quad m$

Table 1-5: Near Real Time Video Streaming Traffic Model

Gaming traffic model

Gaming is a rapidly growing application embedded into communication devices, and thus wireless gaming needs to be considered.

Packet size in gaming traffic is modeled by the Largest Extreme Value distribution. The starting time of a network gaming mobile is uniformly distributed between 0 and 40 m to simulate the random timing relationship between client traffic packet arrival and reverse link frame boundary.

On the uplink, a packet is dropped by the subscriber station if any part of the packet (including HARQ operation) has not started within 160msec of the time the packet entered the subscriber station's buffer. Packet delay of a dropped packet is counted as 180 ms. Currently, understanding is that 50 ms lag is considered excellent quality while 100 ms lag is considered good quality. Ping times above 150 ms are often reported to be intolerable . Outage in wireless gaming is defined as average packet delay greater than 60msec, where average delay is the average of the delay of all packets, including the delay of packets delivered and the delay of packets dropped. Table 1-6 provides the model parameters for wireless gaming .

Component	Distribution		Parameters		PDF
	Downlink	Uplink	Downlink	Uplink	

Initial packet arrival	Uniform	Uniform	a=0, b=40 ms	a=0, b=40 ms	$f(x)=\frac{1}{b-a} \quad a \le x \le b$
Packet arrival time	Extreme	Deterministic	a=55 ms, b=6 ms	40 ms	$f(x)=\frac{1}{b}e^{-\frac{x-a}{b}}e^{-e^{-\frac{x-a}{b}}}, b>0$ $X=a-b\ln(-\ln Y)$ [2], $Y \sim U(0,1)$
Packet size	Extreme	Extreme	a=120 bytes, b = 36 bytes	a=45 bytes, b = 5.7	$f(x)=\frac{1}{b}e^{-\frac{x-a}{b}}e^{-e^{-\frac{x-a}{b}}}, b>0$ $X=a-b\ln(-\ln Y)$ [2], $Y \sim U(0,1)$

Table 1-6: Wireless Gaming Traffic Model

Note:

1] To account for UDP header, 2 was added to the size of the packet size

2] Because packet size has to be integer number of bytes, the largest integer less than or equal to X is used as the actual packet size.

Traffic Mixes and Collected Metrics

Table 1-7 contains traffic mixes that should be used in system evaluations, and collected metric. Definition of output metrics is provided in an accompanying contribution for information.

	VoIP	FTP	HTTP	n.r.t. video	Gaming	Additional Metrics
Full Buffer Voice	**100%**	**0%**	**0%**	**0%**	**0%**	C_{voice}
Full Buffer Data	**0%**	**100%**	**0%**	**0%**	**0%**	C_{data}
Traffic Mix	**30%**	**10%**	**20%**	**20%**	**20%**	
Collected Metrics per Application	$TD_{u,voice}$ $TD_{sec,voice}$ $J_{u,voice}$ O_{voice} $VD_{sec,appl}$	R_{ave} R_{sec} R_u $O_{thpt}(R_{min})$	R_{ave} R_{sec} R_u $O_{http}(R_{min})$	$TD_{u,vid}$ $TD_{sec, vid}$ $J_{u,vid}$ $VD_{sec,vid}$ R_{ave}	$TD_{u,game}$ $TD_{sec,game}$ $J_{u,game}$ $VD_{sec,game}$ R_{ave}	

$\mathbf{R_{sec}}$	$\mathbf{R_{sec}}$
$\mathbf{R_u}$	$\mathbf{R_u}$
$\mathbf{O_{video}(R_{min})}$	$\mathbf{O_{game}(R_{min})}$

Table 1-7: Traffic Mixes