Traffic Analysis and Design of Wireless IP Network

·Cellular Mobile Network is developing since the last decade (→ onyıl) of 20th century it provides,

* different multimedia services
* high bandwidth
* mobility (hareketlilik)
* management of high data traffic of different types of data (voice, video, data)
* real time data (lig maçları) - non real time (youtube'daki videolar)

IP (Internet Protocol) penetrated (→ içine girmek) the world as a result of wide & popular usage of IP Networks for e-mail, web pages.

The first widespread (→ yaygın) application combining video & voice & data was the web browser.

Mobility is provided for the users, by developing much more smaller communication devices.

By 3G, Internet & wireless cellular networks are integrated. This integration was at first created for the hosts connected to LAN's (Local Area Network)

! handover
○ baz istasyonlardan birinden birine geçiş.

The revolution of telecommunication is as:

1) Automatic telephone exchange (at late 1900's)
2) Digitalisation of telecommunication system's (1970's - 1990's)
   → important for transfer of data
3) Integration of circuit switched (connection oriented) telecommunication {delay olmasını engelliyor telde} {kaynaktan hedefe yol belirleniyor, o yolun bütün gereklilikleri sağlanıyor ve connection yapılıyor.}
   and packet switched networks (1990's - 2000's)
   (connectionless) → using bandwidth efficiently
   → {internet üzerinden ses iletimi buna giriyor} {250 - 400 ms aralığı geçince bozuluyor}

These steps are then followed by the mobile systems in development

1) First generation (1G) 1980's (64 K) and analogue voice signals { sadece ses görüşmesi, sms yok }

2) Second generation (2G) 1990's (9600 kbps – 1400 kbps)
ETE (end-to-end) communication is digitalized by integrated services digital network (ISDN) and, Modem based Networks
{ voice data video }

3) 3G 2009's (2Mbps)

Telecommunication Systems

- Switching Part
  - Circuit Switching
    - user 1
    - access network
    - user 2
  - Packet Switching
    - user 3
    - user 4
- Transmission Part
  - Wired Links
  - Wireless Links

Cost for Network Operators

maliyet

1) Equipment and installation costs
2) Operation and maintanance costs
   ↳ işletim ↳ bakım

These costs can be kept at lower levels when all the transmission is carried over a single network for this reason, in early 1990's ATM is developed to provide a single network for different data types and this logic is called "single socket on the wall".

IP supports,

— UDP (user datagram protocol) { güvensiz }

— TCP (transport control protocol) { güvenli } paketin karşıya gönderilmesi anlamında

↳ { OSI 4. katmanda bulunuyor }

However all these services including 32, are still no able to provide as much high bandwidth as wireless fidelity (wi-fi)
{Hiçbirisi wi-fi kadar hızlı değil}

Packet Level Traffic Modeling

During an established internet connection many packets with different sizes have several approaches for modeling the traffic on the packet level.
↳ yaklaşımlar

→ karşılaştırma → genel olan → uygun olan
All of them are based on comparision of emprical and available mathematical models with similar statistical characteristics.

In high priority class we need to model (IP telephone traffic we have A different case with sources with variable bit rate (VBR) and with real time requirements (subclass A_2) or nonrealtime services (subclass A_3 and B)
{network'ün durumuna göre paket gönderiyor}

There's no unique description at the internet traffic due to the great heteregenity of network topologies, protocols and applications.
↳{bazı yerlerde video çok izlenirken, bazı yerlerde mail çok gönderiliyordur, bu yüzden heterojen}

Definition of Wireless IP Networks

A wireless IP Network is an all IP Network with wireless access. All data, signalling and control information are carried by IP packets.

TRAFFIC ISSUES

IPV4

V4	IML	ToS	Total Length
Identification		Fragment Offset	
TTL	Header Checksum		
Source Address			
Destination Address			
Options			

→ for starting point

IPV6

→ for use control information

V6	Priority	Flow Label	
Payload Length		Next Header	Hop Limit
Source Address			
Destination Address			

↳ kim gönderiyor burada yazıyor.
→ nereye gideceği burada yazıyor.
örneğin, 1 → 3 → 5 → 7

One traffic type for all doesn't suit for all applications, and QoS may be extreamly important for some users when resources are scare and shouldn't be wasted the QoS analysis after an ETE analysis of whole network.
↳ end to end

Design of tradional cct. switching network with single traffic class (voice) is carried using traditional approach based on Erlang B. formula.

The formula provides GoS (Grade of Service) which is probability Pb that or new call arriving at the cell group is rejected because of busyness of all services.

Trafiğin yoğunluğundan paket alımı red etmeye Pb denir.

What is IPv4?

IPv4 was the first version of Internet Protocol to be widely used, and accounts for most of today's Internet traffic. There are just over 4 billion IPv4 addresses.

While that is a lot of IP addresses, it is not enough to last forever.

What is IPv6?

IPv6 is a newer numbering system that provides a much larger address pool than IPv4. It was deployed in 1999 and should meet the world's IP addressing needs well into the future.

What is the major difference?

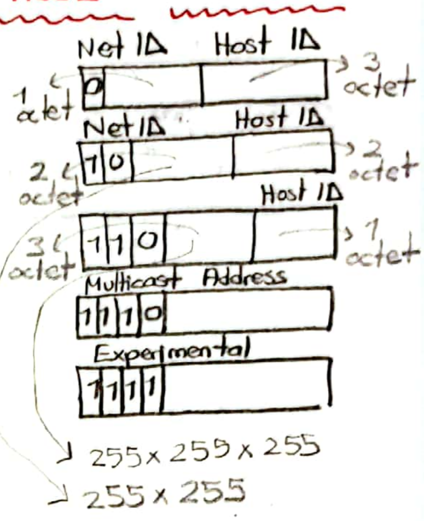
The major difference between IPv4 and IPv6 is the number of IP adresses.

There are 4,294,967,296 IPv4 adresses. In contrast, there are 340,282,366,920,938,463,463,374,607,431,768,211,456 IPv6 adresses.

IP CLASSES

127 → local loop , 1 byte → 1 octet

Class	First Octet Range	Max Hosts	Format
A	1 – 126	16M	
B	128 – 191	64K	
C	192 – 223	254	
D	224 – 239	N/A	
E	240 – 255	N/A	



Differences between IPV4 and IPV6,

IPV4

* Addresses are 32 bits (4 bytes).
* Address (A) resource records in DNS to map host names to IPV4 addresses.
* Header doesn't identify packet flow for QoS handling by routers.
* Both Routers and hosts fragment packets.
* Header includes checksum.
* Configured manually or through DHCP.
* Header includes options.

IPV6

* Addresses are 128 bits (16 bytes)
* Address (AAAA) resource records in DNS to map host names to IPV6 address.
* Header contains flow label field which identifies packet flow for QoS.
* Routers don't support fragmentation.
* No checksum.
* Does not require configuration.
* Optional data is supported as extension headers.
  ↳ genişletme kapsamlı ↳ başlantı

$2^{24} = 2^{10} \cdot 2^{10} \cdot 2^{4} = \underbrace{K.K}_{M}.16$

$= 16M$

internet sitelerinin adreslerinin numaralarının neye karşılık geldiğini bulan sistem → DNS

$B(e,m)$ → The mentioned probability when E Erlang of traffic is offered to m trunks (channels)
(göstermek anlamında)

trunk sayısı ← m → # of sources such as severe of ccts in a group.

i → loop variable

$$Pb = B(E,m) = \frac{\frac{E^m}{m!}}{\sum_{i=0}^{m} \frac{E^i}{i!}}$$

Pb → Probability of Blocking

E → Total amount of Traffic offered in Erlang.

The erlang formula can be in loss systems such as tel. systems and mobile networks that don't have traffic buffering. (paket kayıplarında erlang formula kullanılıyor.)

In wireless networks the traffic distribution (↳ dağılım) and its parameters depend on,

- user mobility
- cell size
- bit rate (throughout)
- network load (birim zamanda yüklenen paket sayısı)
- BS (↳ base station) scheduling (wireless AP) (FIFO en basit hali)
  FIFO ↓ first in first out

In an IP Network there would simultaneously (↳ aynı anda) exist video, voice, data, packets.

Application must be classified as
- realtime
- non real time

} QoS 'e veya priority 'e bakarak karar veriyoruz.

AP (access point)
Bridge
Repeater
} üç özellik AP300'de (kablosuz) var.

Reservation Protocol → gideceğimiz yol üzerindeki paket taşıyacak kanalları önceden belirleyip söyleme.

Some QoS support concept for the internet are IS (Integrated Services), ETE (end-to-end) reservation of resources (connection oriented) MPLS (Multiprotocol Label Switching) to provide unified QoS for different protocols such as IP and ATM.

In next generations when all IP network structure is developed micromobility management will appear Mobile IP protocol is defined as a standard for micromobility (global) but it is not efficient for micromobility

micromobility → kendi içinde gezme

Micromobility includes,

- handover scheme: influences the flow and the ongoing traffic
- routing algorithm
- location control

Definition of Wireless Channel

Amount of bandwidth (BW) allocated to a mobile user at a time (kbps)

- in frequency band
- in timeslots
- access codes

GSM (Global Service Mobile)

1,25 MHz BW (bandwidth) can be used by several subscribe can be spreads as simultaneously by this way a narrowband wideband assigning a unique code to each telephone or data call.

Handover: Baz istasyonları arasında geçiş. Birinde kayıtlıyken diğerinin kopsama alanına girmek.

AVC: Bağlanıp bağlanmamasına karar veren mekanizma (fatura ödememiş bağlanmasın) (avea mı turkcell mi)

cell structure

In communications, different data types have different characteristics and QoS demands statistical characteristics on QoS requirements should be the main parameters for classification of the Aggregate IP traffic.
statistical → istatiksel; Aggregate → combine

★ very important: To provide certain quality within the given constraints on the quality measures IP network must decide to whether reject or accept the calls of the other nodes depending upon traffic conditions.
constraints → kısıtlama, sınırlama; reject → geri çevirmeyeceği

An analytical framework is required in order to dimension and optimize multiclass wireless networks in some cases one should proceed with simulation analysis at traffic senarios.
optimize → en iyi şekilde kullanmak

Design Issues

Wireless Network,

- user mobility is provided
- location based BER and SNR

BER → bit error rate; SNR → signal noise ratio

bu ikisi birbirine ters orantılı

Wired Network,

- fixed user
- BER is independent from the location

fixed → sabit

Some QoS support concept for the internet are IS (integrated services): End to End (ETE) reservation of resources
Multiprotocol Label Switching (MPLS): To provide unified QoS for different protocols such as IP and ATM.
unified → birleşim
Differential Services (Diff. Serv.): Specifies per hop behaviours instead of ETE services.

Wireless LAN,

- internet principles
- BE class support only

BE → best effort

3G,

- cct switching + packet switch by 2G + IP accesability
- Voice over IP'de cct switching + packet switch var.

connection oriented ve connectionless ters orantılı.

Connection oriented → kaynağın hedefe gideceği yolun belirlenmesi, gideceği yolla ilgileniyor. Gidip gitmediğini kontrol ediyor. (TCP)
Connectionless → Paketin gittiği yolla ilgilenmiyor. Gidip gitmediğine bakmıyor. (UDP)

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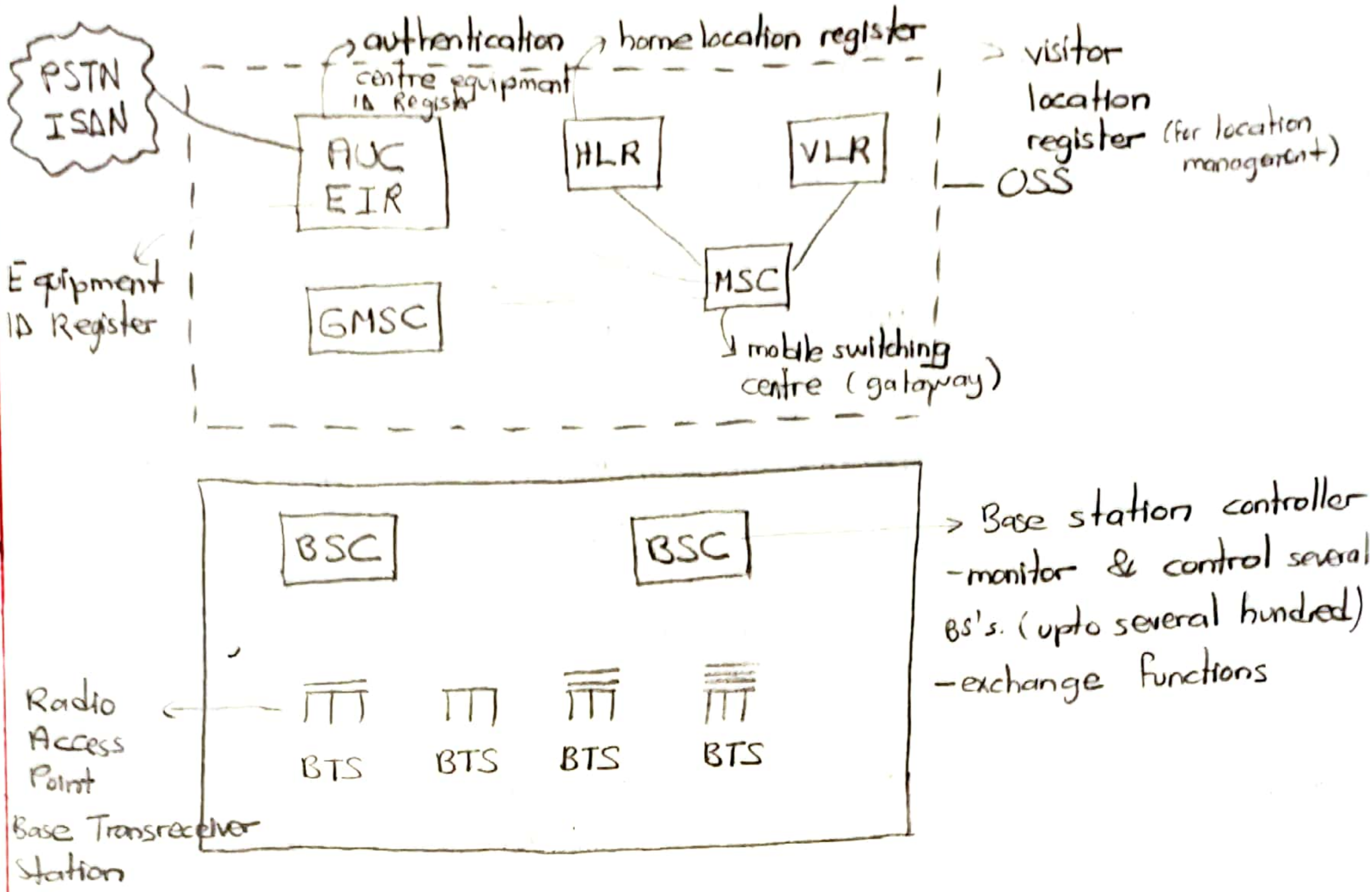
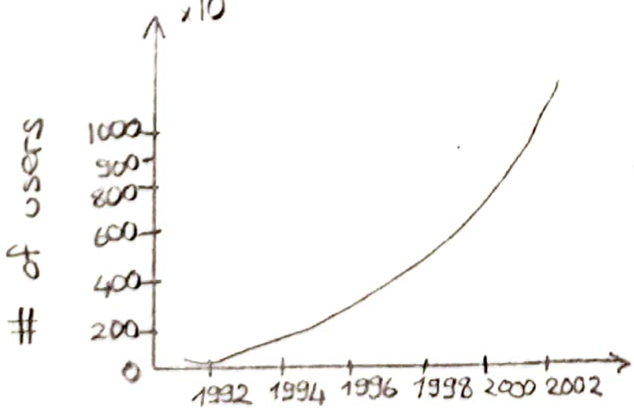


HLR : Kendi baz istasyonundaki kişileri tutuyor.
VLR : Geçici kişileri kaydediyor.
MSC : Baz istasyonlarında switching yapıyor.
PSTN : Public Switching Network
BSC : Yüzlerce baz istasyonunu kontrol ediyor.

3G Wireless Mobile Communications and Beyond

The figure below illustrates the expanantial increase in the number of mobile subscribers between years 1992 - 2002 from the



beginning of internet (ARPANET) the number of users and host computers are always double a each year of course total internet traffic increases faster than the # of hosts in the internet.

As a technology internet is based on IP which is transparent enough to support transmission of different data types.

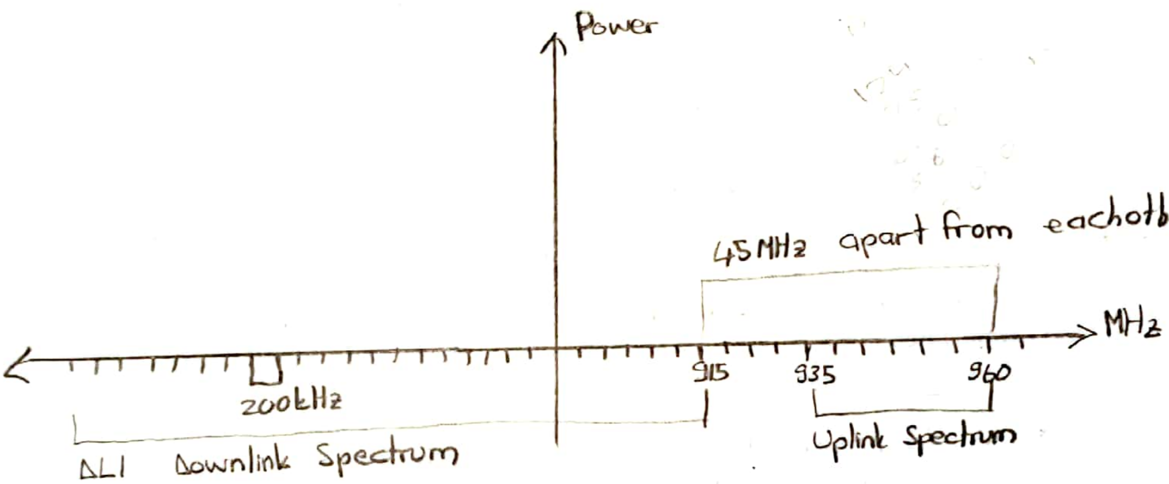
2G Mobile Networks

The first trials (denemeler) with 2G began with GSM in 1994 then by april 2002 there were about 400 GSM operators around the world.

ISDN applied to GSM system and it was then a fully digital system GSM is a combination of IDMA & IDMA methods.

GSM 900 was the first digital one and used. 900 MHz frequencyband with 25 MHz frequency spectrum, using 124 carrier frequencies of 200 kHz each. Each frequency is divided into 8 time slots by use of TDMA technique.

Carries 45 MHz apart (bölüm) from each other are used to be able to provide 2 way communication. This carrier pair is called absolute radio frequency channel number (ARFCN)



These frequency carries are also used by several nodes by multiplexing with TDMA technique. The TDMA frame structure is as shown below

← frequency duration = 4,61ms →

TS1	TS2	TS3	TS4	TS5	TS6	TS7	TS8

loop of 36,88ms

$\frac{36,88\ ms}{8}$ ⇒ her TS kaç ms bulunz.

Frame duration 4,61ms

— GSM (Global Service Mobile)

— AMPS $\xrightarrow{\text{developed}}$ D-AMPS (Digital Advanced Mobile Phone Systems)

TS1	TS2	TS3

→ Known as TDMA based IS-54 standard

30kHz

IS54 $\xrightarrow[\text{+SMS}]{\text{developed}}$ IS-136 standard

— CDMA (Code Division Multiple Access) → known as IS-95 (1993)

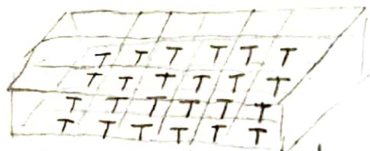
GSM → TDMA / FDMA

TDMA: time division multip. access

FDMA: frequency division multip. access

Bu ikisinin çaresiz kaldığı zamanda CDMA işin içine giriyor. Her kullanıcıya bir kod atanıyor.

Otopark örneği gibi düşünüyoruz



Arabaları tek çift diye ayırıyoruz. Gündüzleri tekler akşamları çiftler (zamanı ayırdık) Aşağı kat dolduğunda üst katıda aynı hizada kullanıyoruz. (frekansı bölüyoruz)

EVOLUTION FROM 2G to 3G

Rate of 2G was too slow for many internet services. Then 2G+ is generated from 2G and other TDMA based techniques. HSCSD (High speed cct. switched data) and GPRS are in this group EDGE (Enhanced Data Rates for Digital Evolution) is also classified in this group HSCSD is a software upgrade to GSM networks In HSCSD a single user can use multiple channels simultaneously (upto 4 of 8 in a single TDMA time slot)

4'e kadar

This means 14.4 kbps x 4 = 57,6 kbps ≅ Dial up modem connection

{ 3G > 2,5G > 2G (GSM)   2,5G => E/H   E>H }

GPRS – Tracing the way to mobile internet

* Data Rates of 2G were slow for internet usage
* cct switched networks were expensive for bursty internet traffic
  → (each user uses single line, so expensive)
* paket switched networks needed to be designed for statistical multiplexing and GPRS appeared as a development of GSM, this was the first step towards integration of internet and Mobile Cellular Networks.

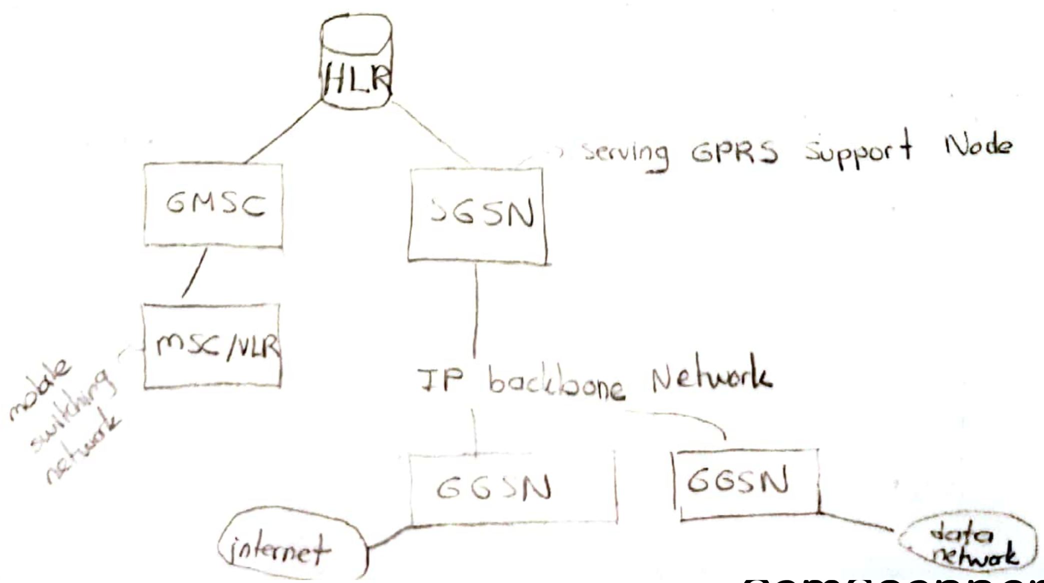
— Packet based
— Can use 1 to 8 time slots in a TDMA systems.
— Can use HW and SW upgrades to existing GSM.
  HW → hardware, SW → software
— 2 new nodes are added SGSN (Serving GPRS Support Node) and GGSN (Gateway GPRS support node)

SGSN (Responsable of packet delivering and charging

GGSN : Interface between GPRS network & external packet based networks, converts protocol datapackets (PDP) address and GSM addresses to each other.

internetin GSM den girişi nasıl oluyor.

GPS Network Architecture



GPRS supports different QoS profiles for variaty of data types such as realtime multimedia, nonrealtime multimedia, www, FTP, e-mail

QoS is defined in terms of

— service precedence (servis önceliği)
  ↳ öncelik
— reliability (güvenilirlik)
— delay (if delay increases, communication quality decreases) (gecikme)
— throughput (the number of bytes transmitted)
  (veri hacmi) (eğer bu sağlanıyorsa kullanıcı delay ve diğerleriyle ilgilenmez.)
  (diğer üçü buna hizmet eder)

(Yayına o anda verilen bir eski film veya maçı internette izlemek realtime olur. illa o an maçın oynanması gerekmez.)

EDGE (Enhanced Data Rates for GSM Evolution)

Higher data rates using existing 2G systems.
Uses a new modulation scheme 8-psk in addition to GMSK (Gaussian Minimum Shift Keying) in GSM / GPRS. It enables data rates upto 384 kbps.

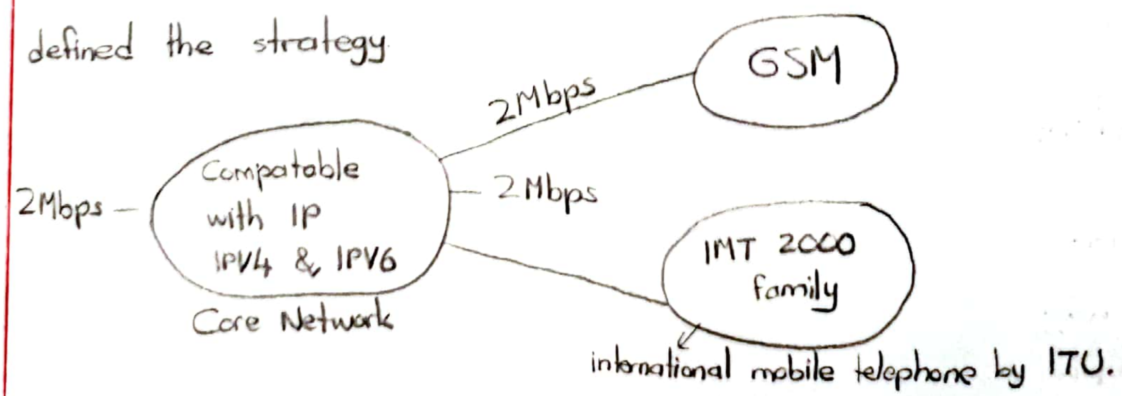
EDGE — for wide area coverage
UMTS — urban hotspots (kentsel kablosuz erişim alanı)
  ↳ yayın alanı

Third Generation Mobile Networks

Converges the standards in 2G: CDMA + GSM + TDMA improve data rate with packet based structures.

UMTS

UMTS is the ETSI candidate for 3G. This standartization defined the strategy



Supports services with different QoS characteristics.
Four QoS classes are defined for UMTS, these are;

Real Time Traffic
- karşılıklı konuşuyoruz.
- Conversational class
- streaming class (one way)
- film izliyoruz

Non Realtime
- bankalar, gazete okumak
- interactive class
- background class
- veritabanının güncellenmesi

referred as traffic classes

ayırt edici

Main distinguishing parameter is defining the realtime services and the delay. According to this,

Conversational class is very delay-sensitive traffic.
Background class is the most delay-insensitive traffic.
Streaming class is one way realtime transportation.

akış durduran işlem

	Conversation Class	Streaming Class	Interactive Class	Background Class
Def.:	cct-switched telephony	real-time video stream	Apps. requesting data from remote (e.g. server)	Receiving and sending by the computer
Ex:	real time VOIP communication	real time video stream	web access database access	e-mail SMS

Traffic Class	Conversation	Streaming	Interactive	Background
Maksimum Bit Rate	X	X	X	X
Guaranteed Bit Rate	X	X		
Delivery Order	X	X	X	X
Maximum SDU (Service Data Unit)	X	X	X	X
BER	X	X	X	X
SDU Error Ratio	X	X	X	X
Transfer Delay	X	X		
Traffic Handling Priority	X X X	X X	X	

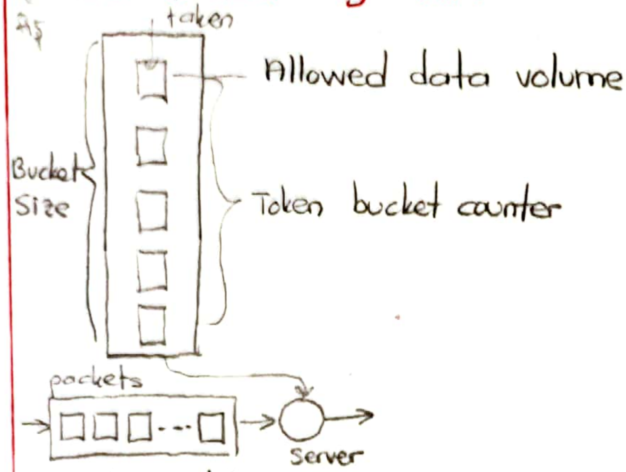
Traffic ↔ Max Data Rate
↓
Guaranteed Bit Rate

Taken rate → Min bit rate (Allowed Data Volume)

Bucket size → Max 500 size

Takens are generated according to traffic contract and stored in a taken bucket. If the taken bucket is full they will be discarded if TBC (Token Bucket Counter) is greater than incomming packet length the traffic is conferment. Else the packet is marked as non-conferment.

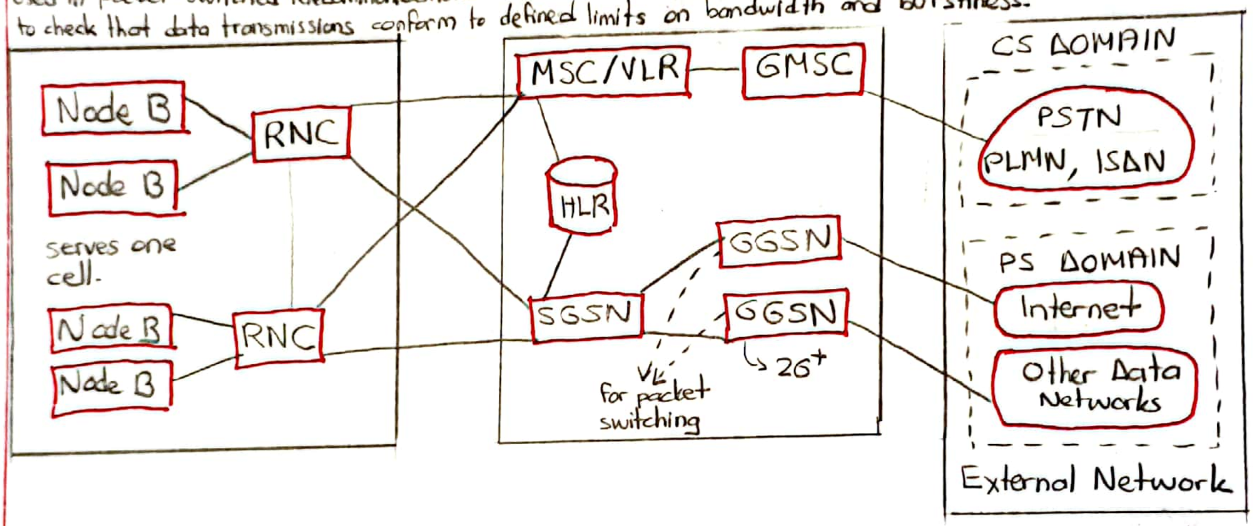
Token Bucket Algorithm



The token bucket is an algorithm used in packet switched telecommunications networks to check that data transmissions conform to defined limits on bandwidth and burstiness.

UMTS Architecture

There are two triviol domains in UMTS, these are; user equipment (UE) and the infrastructure domain. UE is the users used to access to UMTS services. UE includes identity mobile equipment, mobile equipment includes software and hardware groups for radio communication.



- GGSN : Gateway GPRS support node
- SGSN : Serving GPRS support node
- PSTN : Public switched telephone network
- GMSC : Gateway mobile switching center
- HLR : Home Location Register
- RNC : Radio Network Controller

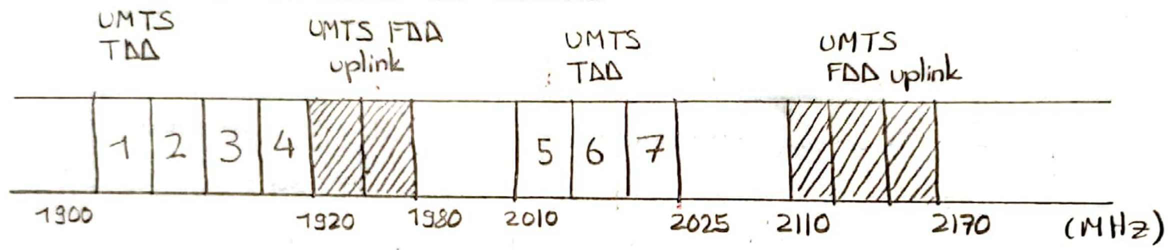
UMTS-3G Network Architecture

Comparison of 2G and 3G Mobile Networks

2G'den ekstra bunlar var. Kaldırılan bişi yok. Yani internet eklenmiş.

Network	Second Generation	$2G^+$	Third Generation
Core Network	MSC / VLR, GMSC, HLR (MSC ↳ mobile switching center); AuC, EIR (AuC ↳ Autotantication Center; EIR → equipment id register)	MSC / VLR, GMSC, SGSN, GGSN, HLR, AVC, EIR	3G-MSC / ULR, 3G-GMSC, 3G-SGSN, 3G-GGSN, HLR, AUR, EIR. $2G^+$'dan farklı bişi yok. Sadece hızı arttı.
Radio Access Network	BSC, BTS, MS (BSC ↳ base station controller; BTS ↳ base transmitter; MS ↳ mobile switching)	BSC, BTS, MS	RNC, accesnode, mobile station (RNC ↳ radio network controller)
Mobile Station	Voice only terminals	user friendly terminals	voice, data, video

FREQUENCY BANDS OF UNITS



Hocanın kitabı siteye koyması → uplink
Öğrencinin siteden kitabı indirmesi → downlink

WCDMA

in FDMA / TDMA common channel space is partitioned in orthogonal single user subchannels without overlapping (üst üste binme) when we have bursty traffic problem occurs. Systems always have to wait for cilent durations (not speaking or web browsing)

WCDMA supports upto 2Mbps data rates utilising variable spreading factor and multicode links. (Aynı frekans ve aynı zamanda oluyor WCDMA)

If user data remains constant, it is transmitted using 10ms fixed frames

By variable spreading factor actual carrier bandwidth is addressed between 44 kHz and 5kHz by using grid of 200kHz.

Some network features are specific to WCDMA or CDMA soft handover. Connection to more than 1BS is allowed. By soft handover, handover is done between these base stations. By softer handover, handover is done between the cells belonging to same BS. bir baz istasyonunda iki cell varsa birinden ötekine geçme softer handover.

To switch from TDD to FDD or FDD to TDD only hard handover is possible.

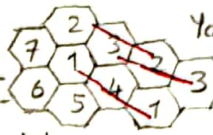
Multipath Reception:

The receivers also allow to decode multiple signals travelled through different paths of reflection.

Frequency Reuse of 1:

yeniden kullanım

Avantaj. Günkü normalde 7 farklı frekans kullanılıyor.



Yanına gelen 7'lik cell'de bu frekansları kullanıyor. Ama tek bir frekans olursa aynı anda kullanılır ve trafik olmaz. Öbür türlü karışıyor.

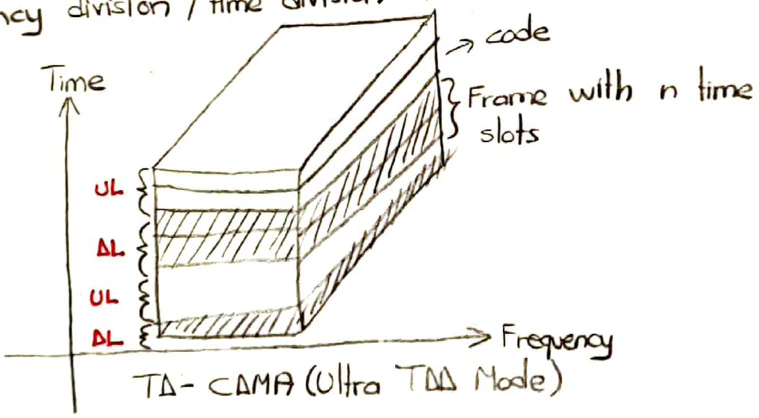
Same frequency band can be used in neighbouring cell, so not frequency planning is required.

TD - CDMA (aynı frekansta herkes uplink ve downlink yapabiliyor.)

It uses same frequency carrier for uplink and downlink with Time Division Dublexing.

Uplink and downlink time slots are grouped into sequences.

It can assign any time slot for either uplink or downlink. It uses frequency division / time division and code division simultaneously as seen below



TD- CDMA (Ultra TDD Mode)

Soft handover → hocanın sınıfta iki öğrenciyle ayrı ayrı konuşması. İki öğrenci de ekteki konuşmayı duyuyor.

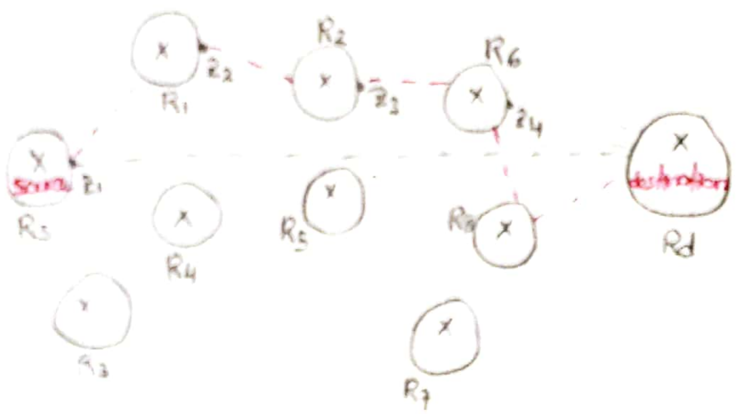
Softer handover → öğrenciler sınıfta hoca sınıfın dışında bizimle konuşması

Hard handover → öğrenciler bir sınıfta Hoca ...

Routing and Relay Selection Algorithms

Relay Selection Algorithms

Shortest path Routing Algorithm



$$Path_{min} = \min\left[d(T_x, R_1) + d(Rs_{n+1}, Rd) + \sum_{n=1}^{hc} d(Rs_n, Rs_{n+1})\right]$$

↳ n^{th} relay of the S Route.

FASTEST PATH

$$C = B \log_2 \left(1 + \frac{S}{N}\right)$$

$$S = P \cdot d^{-\alpha}$$

$$C = B \log_2 \left(1 + \frac{P}{N} d^{-\alpha}\right)$$

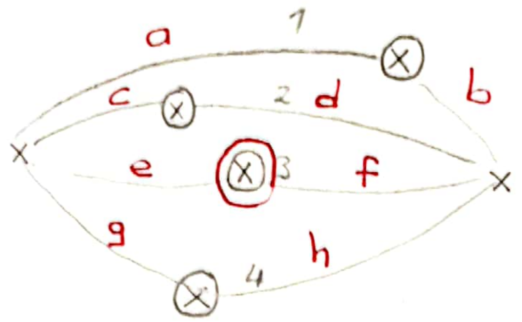
$$z = (1-\gamma) U_i + \gamma U_d$$

$$\gamma = \frac{X_s}{d(\text{Source, Destination})}$$

$$X_s \approx \left(\frac{P}{N(e^{\alpha}-1)}\right)^{\frac{1}{\alpha}}$$

$$Path_{min} = \min\left[\left(d(T_x, R_s') + d(R_s^{hcs-1}, R_s) + \sum_{n=1}^{hcs-2} d(R_s^n, R_s^{n+1})\right)\right]$$

Min-Max Distance



	max	min
1.	a	
2.	d	
3.	e	*
4.	h	

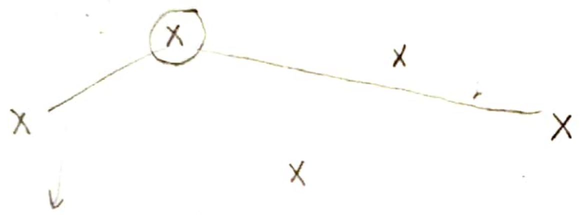
↳ her yolun maksimumunu buluyoruz. Daha sonrada bu maksimumlar arasından en kısayı seçiyoruz.

sort a-b, c-d, e-f, g-h

find the minimum one select the relay $Path_{min-max} = min$

$$= \min\left(\max\left(d(T_x, R_s^1), d(R_s^1, R_s^2), ----, d(R_s^{hc-1}, R_s)\right)\right)$$

Closest Node To The Source



for having small error probability

$path = \min\left(d(R_n, T_x)\right)$ will be selected.

Decode forward'ın amplifier'a göre dezavantajı, gelen sinyal belli bir thresholdun altındaysa ne olduğunu anlamaya çalışıp yanlış bile olsa tahmin ettiği şeklinde gönderir. Mesela, "sağ" kelimesindeki a bozuk olsun. Onu tahmin etmeye çalışıp "seç" şeklinde yanlış şekildede gönderebilir. Min-Max işte bu durumda bozulmamayı sağlıyor. Yani decode kullanıyor ve uzak olan yolları seçmemeye çalışıyor.

Relay Selection According to Power Threshold

→ freespace olduğunu anlıyoruz.

$$P_r(R_n) = P_t\left[\frac{c}{d(T_x, R_n)(4\pi f)}\right]^{2} \geqslant \text{Powerthreshold}$$

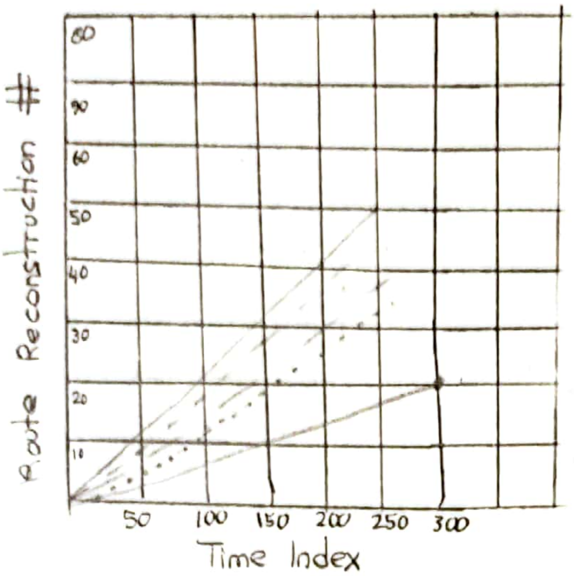
Powerthreshold snr hassasiyetine göre değişir.

with smallest $P_r(R_n)$ will be selected

Relay selection according to Pathloss where the free space loss can be calculated by

$$FSPL(dB) = 20\log_{10}\left(\frac{4\pi f d}{c}\right)$$
$$= 20\log_{10} d + 20\log_{10} f + 20\log_{10}\left(\frac{4\pi}{c}\right)$$
$$= 20\log_{10} d + 20\log_{10} f - 147.56$$

Comparison of Relay Selection Algorithms



Path Loss
Closest to Transmitter
Minmax
Shortest Path
Power Threshold
Assoc. Based Routing (ABR)

Long-Life Path Selection Algorithms

ABR

birleşme

Associativity Based Routing uses tables to maintain the network access and neighborhood relations. Each mobile node in the network structure works cognitively for the benefits of all network.

↳ kavramsal olarak

They all keep their own table and keep the records of Associativity Ticks of all their neighbours, availability and Associativity Tick Threshold Value.

Associativity Tick: Used for illustrating the duration of each neighbour that it stayed near the node. For this purpose each node continiously transmits "I'm here" messages to all, it's neighbours. When it leaves the node the messages will be automatically stopped. These messages are called Associativity Ticks.

Availability: The field in the table, which has value of one if the corresponding neighbour continious transmitting AT. messages and AT number is greater than AT threshold value. else it will be zero. The transmitter uses this value while seting up the routes.

AT threshold : The value that a node will be said to be available if number AT is greater than a threshold value. This threshold value is calculated specifically and can be changed by the designer.

Aslında ATAABR için

AT Table:

Amaç, en uzun ömürlü rotayı bulmak. Bu yüzden komşular içerisinden kendisine en uzun süredir komşu olan ve en geniş çevreye sahip olup komşularınında hedefe giden yola sahip olanları seçer.

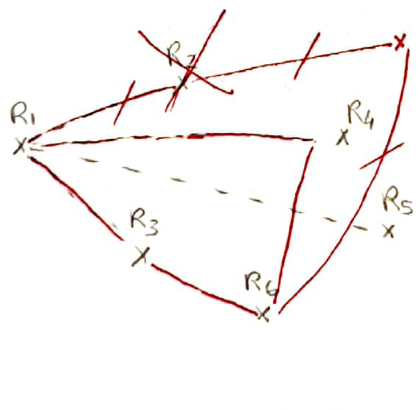
Table of R1	R1	R2	R3	R4	R5	R6
Availability	–	1	1	1	1	0
Number of AT	–	3	2	4	7	0
AT Threshold	–	16/6	12/6	14/6	15/6	5/6

→ nekadar süredir komşun olduğunu gösteriyor. (R5 7 sn'dir komşusu)

↳ 3, 16/6'dan büyük olduğu için bağlantı kurabiliyoruz.
↳ 2, 12/6'ya eşit olduğu için " kuramıyoruz. / Bazen kurabilir.
↳ 4, 14/6'dan büyük olduğu için bağlantı kurabiliyoruz.

Table of R4	R1	R2	R3	R4	R5	R6
Availability	1	1	1	–	0	0
Number of AT	4	5	2	–	0	3
AT Threshold	16/6	16/6	12/6	–	15/6	5/6

$$R_{1\,threshold} = \left(\frac{3+2+4+7+0}{\#\ of\ nodes}\right) = \frac{16}{6}$$



Önce ✓'lı yolları seçtik. Onlar arasından R6'ya direk bağlananları bulduk. Onlardan herhangi birini seçiyor. R2, R6'ya gitmeden başka birtanesi üzerinden gittiği için onu eledik.

EABR (Enhanced Associativity Based Routing)

EABR algorithm is an extansion of ABR, only with the difference that destination has an active role in making decisions for route reconstructions in EABR, where it is passive in ABR.

AEABR (Alternative Enhancement in ABR)

Main working principle of AEABR algorithm looks like a combination of ABR and EABR.

Table R4	R1	R2	R3	R4	R5	R6
Availibility	1	1	1	—	0	0
# of AT	4	5	2	—	0	0
AT Threshold	$\frac{16}{6}$	$\frac{16}{6}$	$\frac{12}{6}$	—	$\frac{15}{6}$	$\frac{5}{6}$
Old Power	1,2	1,8	1,19	—	0	1
New Power	1,2	1,9	1,34	—	0	1

It requires a modification where there exists more than one path having the same number of hops in the set of possible routes determined by EABR algorithm.

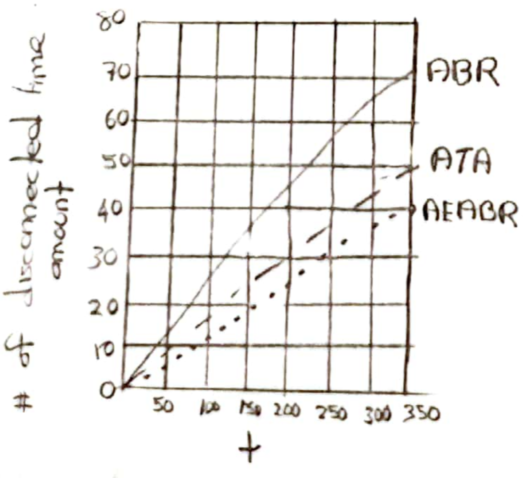
At messages can only be received if and only if the received power is greater than a predefined threshold power value.

$$Path_{pw} = \left(P_w(T_x, R_s^1) + P_w(R_s^{hcs-1}, R_x) + \sum_{n=1}^{hcs-2} P_w(R_s^n + R_s^{n+1}) \right) / hcs$$

ATAABR (Associativity Tick Averaged Associativity Based Routing)

Main working principle of ATAABR is also an extension of ABR with ATAABR, at firsts the paths are selected for which all nodes are linked to eachother such that; number of AT's are greater than calculated AT threshold value and among the set of these selected paths, the which have min hop count are selected for which average of the AT values in each route has the max value.

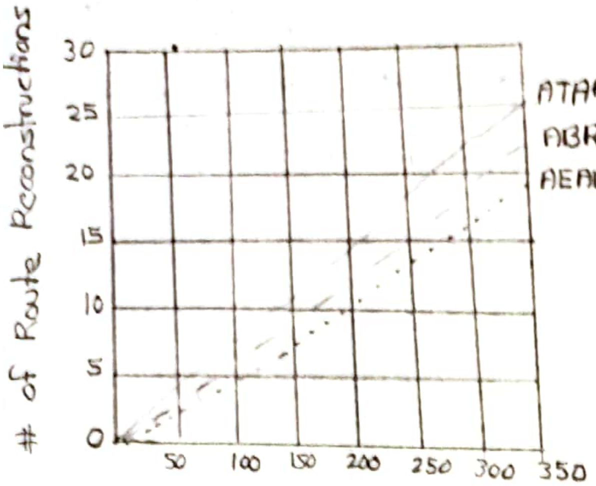
$$Path_{ATAABR} \left(\left(AT(T_x, R_s^1) + At(R_s^{hc-1}, R_s) + \sum_{n=1}^{hcs-2} At(R_s^n, R_s^{n-1}) \right) \right) / hcs$$



Rastgele seçen → ABR

En geniş çevreyi seçen → ATAABR

Power'a bakan (uzaklık değişimine yani)
↳ AEABR



The network is too crowded ⇒ ATAABR and ABR

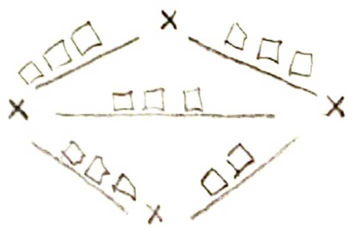
Data being transmitted in the network is mostly real time transmission ⇒ AEABR and ATAABR

We have limited bandwidth and bandwith contraints ⇒ Fastest Path, ABR ATAABR

Reliability is desired ⇒ Fastest Path ABR, AEABR ATAABR

IP

* **IPV4:** The internet is built over IPV4 which is designed for interconnected packet switched networks



— Datagrams (connectionless delivery)

IPV6 is created to overcome the IP address starvation of IPV4.

in IPV4,

— Datagrams are transmitted by fragmantation (bölümlendirme) of long ones

— Transmission is provided from source to destination

IP Function

- Addressing → carried in ID nodes in 4-Bytes — used for independently routing the packets towards destination by routers
- Fragmantation

Header structures for IPV4 and IPV6 were already drawn before datagram transmission is not reliable and there are no acknowledgements.

Checksum: Error check of the header
Each router first chucks the error then reduces the TTL value by one.

Identification: Includes identifying value assigned to assembly the fragments of the datagram
For reliability we need extra functions such as TCP.

***IPV6:**

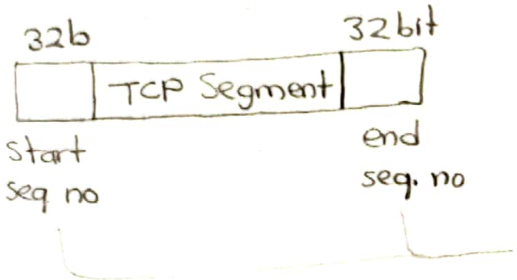
- Smaller address space is enlarged (32 bits → ?)
- Addressing capabilities are Expanded
  multicast + anycast
  added to send message to anyone
- uyumlu
  Compatible with IPV4
- Flow labeling capabilities for the packets belonging to some flow
- Autantication and privacy capabilities
- Since lower layers already use Error control mechanisms checksum is removed in IPV6.
- Traffic class field corresponds to TOS field in IPV4.
- Some fields of IPV4 are dropped, because IPV6 has higher header redundancy due to increased address length
- more IP addresses
  flow control mechanisms
  authantication mechanisms } added

TCP (Transmission Control Protocol) (Kayıpsız veri gönderimi sağlayabilmek için TCP protokolü yazılmıştır.)

- Used for reliable data transportation in internet (güvenli)
- Most of the internet services are based on TCP/IP such as www, FTP, telnet...

— Powers %95 of bytes %85-95 of all IP packets %75-85 of all flows in the internet, where the rest mainly use UDP (used for real time data transmission)

— Based on acknowladgements of succesfully received packets (Automatic repeat request (ARR) family member)



# of bytes transmission successfully

TCP Mechanisms

— Reaction in case of packet losses because of congestions or routers/switches.
tepki
silsilelik

— Retransmits only the missing segments
— Retransmits if ack is not received in predefined timeout
— Decreases the data rate in case of increasing congession
— Adaptive size is used for the windows size and indirectly adaptive data rate.
— products are Retransmitted
Timer driven retransmission (By aknowledges)
Data driven retransmission (Fast recovery and Fast Retransmission)

Fast Retransmission

Forces TCP to wait for a small number of duplicate acknowledgements to be received at transmitter.

Fast Recovery

Used with Fast retransmission for invocation of congestion avoidance. The algorithm is as:
↳ yürütme
↳ sakınma

After receiving three duplicated ACKs in a row:
→ tekrarlamalı

1. Set ssthresh to half the current send window
2. Retransmit the missing segment
3. Set cwnd = sshtresh + 3.
4. Each time the same duplicated ACK arrives, set cwnd++. Transmit a new packet, if allowed by cwnd.
5. If a non-duplicated ACK arrives, then set cwnd = ssthresh and continue with a linear increase of the cwnd

TCP SIMULATION HERE

Stream Control Transmission Protocol (SCTP)

— It is designed to transport signaling messages from the PSTN over IP networks but can be used in broader applications too.

— Targeted to create internet equivalent to ITU-1 Signaling System 7 (SS7)

— Originally called Common Signalling Transport Protocol

— Has similarities with TCP

— Reliable transport service
→ güvenilir

— Session oriented mechanism

— TCP Friendly connection and flow control

— Flow and congestion control mechanisms follow TCP algorithms.
↳ yoğunluk

i.e. Slow start after retransmission
congession avoidance
fast recovery → iyileşme, düzeltme
fast retransmit

Single SCTP endpoint supports multiple addresses (multihoming) (increase survivability of SCTP sessions in case of network failures.

Unlike TCP which assumes a single stream of data, SCTP allows data to be partitioned into multiple streams so other streams will not be affected in case of message lose in a stream.

SCTP is considered as an alternative to provide signaling over IP core network in UMTS in preference to TCP and to SS7 used in cct-sw. network.

Ekstra

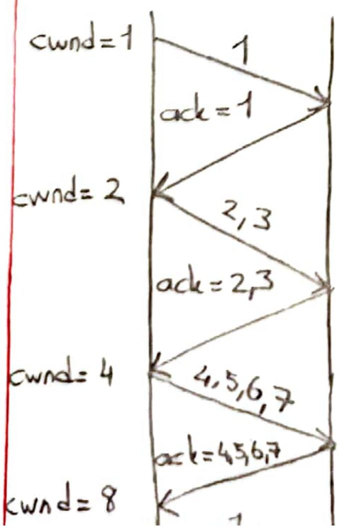
Fast Recovery

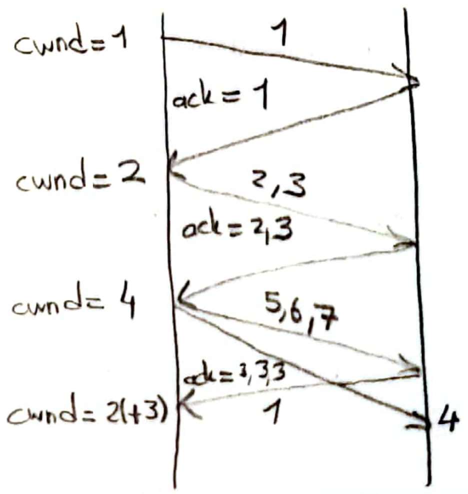
sstresh = cwnd /2 → fast recovery

Q: Consider a TCP flow implementing fast recovery. Assume the end-to-end delay of each packet is d1 in both directions.
Suppose the 4th packet transmitted by the sender experiences a delay of d2 > d1. Assume the first packet is transmitted at time t. What is the congestion window size at time $t + 6 * d1 + \varepsilon$?
Note: Ignore the time it takes to send/receive a packet and assume that the receiver sends an ack as soon as it receives a packet, and the sender sends the next packet as soon as it gets an ack. Assume there are no packets losses.

A: i) d1 = d2 ; cwnd = 8

ii) d1 < d2 ; cwnd = 2 or 5 (both answers were considered correct)





Ekstra TCP / IP

• Kullanım olarak iki katmanlı bir haberleşme protokolüdür.

• Üst katman TCP (Transfer Control Protocol) verinin iletiminden önce paketlere ayrılmasını ve alıcıda bu paketlerin yeniden düzgün bir şekilde birleştirilmesini sağlar.

• Alt katman IP (Internel Protocol) ise, iletilen paketlerin istenilen ağ adresine yönlendirilmesini kontrol eder.

TCP / IP Katmanları

* Application Layer (Uygulama Katmanı) ⇒ farklı sunucular üzerindeki süreç ve uygulamalar arasında iletişimi sağlar.
(HTTP, FTP, SMTP, SSH, POP3, TLS/SSL, DNS, etc.)

* Transport Layer (Taşıma Katmanı) ⇒ Noktadan noktaya veri akışını sağlar.
(TCP, UDP, etc.)

* Network Layer (Internet Katmanı) ⇒ Router'lar ile birbirine bağlanmış ağlar boyunca verinin kaynaktan hedefe yönlendirilmesini sağlar.
(IP, ICMP, IGMP, etc.)

* Link Layer (Ağ Erişim Katmanı) ⇒ Uç sistem ile alt ağ arasındaki lojik arabirime ilişkin katmandır.
(Ethernet, WiFi, PPP, SLIP, etc.)

Physical Layer (Fiziksel Katman) ⇒ İletişim ortamının karakteristik özelliklerini, sinyalleşme hızını ve kodlama şemasını belirler.

TCP/IP ile OSI Arasındaki Farklar

• TCP/IP haberleşme görevini karmaşık bir iş olarak niteleyerek daha basit alt görevlere böler. Her bir alt görev diğer alt görevler için belirli servisler sunar ve diğer alt görevlerin servislerini kullanır. OSI modeli de aynı kavramı kullanır, ancak OSI modelinde her bir katmandaki protokollerin özellikleri ve birbirleri ile ilişkileri kesin bir dille tanımlanmıştır. Bu özellik OSI modeli ile çalışmayı daha verimli kılar.

• OSI modelinde katmanların görevlerinin kesin bir şekilde belirlenmiş olması yeni bir protokol geliştirmeyi kimi zaman güçleştirebilir.
TCP/IP ise böyle bir kısıtlama getirmediğinden, gerektiğinde yeni bir protokol mevcut katmanlar arasına rahatlıkla yerleştirilebilir.

- OSI modelinde gerekmeyen bir katmanın kullanılmaması gibi esnek bir yapıya izin verilmemektedir.

TCP/IP ise katı kurallarla tanımlı olmadığından gereksinim duyulmayan katmanların kullanılmamasına izin verir. Örneğin uygulama katmanında olmasına rağmen doğrudan IP üzerinden kullanılabilen protokoller mevcuttur.

OSI	TCP/IP
Aplication	Application
Presentation	
Session	
Transport	Transport (host-to-host)
Network	Internet
Link	Network Access
Physical	Physical

Difference between OSI and TCP/IP

OSI

1) It has 7 layers.
2) Transport layer gurantees delivery of packets.
3) Horizontal approach.
4) Seperate presentation layer.
5) Seperate session layer.
6) Network layer provides both connectionless and connection oriented services.
7) It defines the services, interfaces and protocols very clearly and makes a clear distinction between them.
8) The protocol are better hidden and can be easily replaced as the technology changes
9) OSI truly is a general model
10) It has a problem of protocol filtering into a model

TCP/IP

1) Has 4 layers.
2) Transport layer does not gurantees delivery of packets.
3) Vertical approach.
4) No session layer, characteristics are provided by transport layer
5) No presentation layer, characteristics are provided by application layer
6) Network layer provides only connection less services
7) It does not clearly distinguishes between service interface and protocols
8) It is not easy to replace the protocols
9) TCP/IP can not be used for any other application
10) The model does not fit any protocol stack.

→ MPLS teknolojisi en basit olarak OSI 2. katmanındaki anahtarlama ve OSI 3. katmanındaki yönlendirme işlemlerinin entegre edilmesi olarak açıklanabilir.

MPLS (Multiprotocal Label Switching)

→ iç içe olma

— Each packet has an MPLS header encapsulated between the link layer and Network Layer headers

— Router with MPLS capability is called label switching router (LSR)

— Routers analyse the header only while forwarding the packet so MPLS is said to be a a packet forwarding scheme

— Networks other than IP are also supported

ingress router : the router that adds the header

eggress router : the router that extracts the header

↳ çıkış noktası ↳ alınmak

MPLS'in asıl olma amacı packet forward'ı yapmak.

20 bits label value	3 bits serv. class	1 bit label stack	8bits TTL
(kime göndereceğini buradan karar veriyor.)	(voice - video - data)		packetin ne zaman öldürüleceğine, bitirileceğine karar veriyor.

L2 Header	MPLS Reader	IP packet

— MPLS uses protocols to distribute labels within the domain to set up LSP's between ingress and eggress points. (MPLS'te veri iletimi etiket anahtarlamalı yollar üzerinden gerçekleşir.)

→ etiketleri yönlendirir

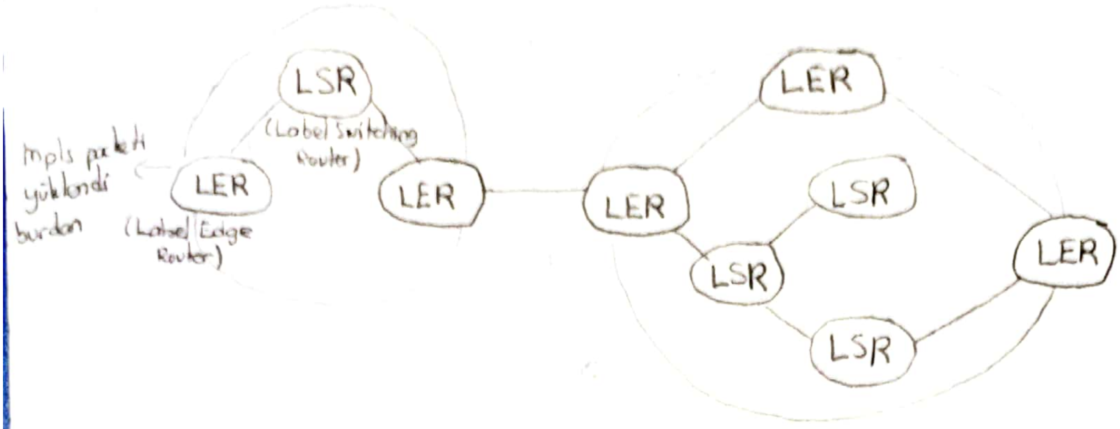
— MPLS uses RSVP or LDP (Label Distribution Protocal for this purpose)

↳ (reservation protocol)

— When LSR receivers a labeled packet, uses the label as the Index to look up the forwarding table.

— Each pack gets an MPLS Label at the enterence of an MPLS domain

(RSVP → önce hangi packetin gitceğine karar veriyor bunun için service class'a bakıyor)



— The main advantages of MLPS are

* Faster Forwarding
* Efficient tunneling of packets (reservasyon yapılmış bir yol oluşturuyor)

— MPLS can be applied to wireless IP networks too.

IS (Integrated Services)

— Integrated Services is defined to support reeltime services in the internet

— Assumes that resources are reserved for every flow requiring QoS at every router hop in the path between the sender and the receiver.

— End to end path is established to support per-flow (flow başına) traffic management (using RSVP signaling). By this way intermediate routers don't store routing information for each flow.

— Provides 2 additional QoS classes.

1) Guaranteed Service

— used for applications requiring bounded end to end (ETE) delay

— Örneğin ses, 250 ms - 400 ms arasında olmalı. Bu sınırları aşmicak şekilde olmasını sağlayan servis guaranteed service.

delay

- fixed (değişmeyen delay)
- queuing (değişen delay) → determined by Guaranteed Service

2) Controlled Load Service

- used for applications requiring reliable and enhanced best efford service.

→ giriş

- uses admission control to Assume that this service is received even when the network element fails. (herhangi bir sıkıntıda da paketin gönderileceği garantisini veriyor.)

2.1) RSVP (Reservation Protocol)

makes reservation in the routers along the path of the packets from service to destination.

RSVP

Hard State — Soft State

→ paketler önceden belirlenen yoldan gönderilir.

* Hard State: Connection oriented, all packets go through the same intermediate nodes

* Soft State: Connectionless, the reservation for a specific flow is saved in a cache at intermediate routers, updated periodically.

→ such a broadcast ↳ packeti herkese gönderme

= allow both unicast and multicast reservations.
↳ packeti bir kişiye gönderme ↳ packeti iki kişiye gönderme

So, the wireless access technology must make the reservations.

2.2) Admission Control Mechanism (reservasyon yapılıp yapılmayacağına karar veriyor.)

Decides whether a request for resources can be granted

Local accept / reject desicion is made at each node

2.3) Classifier (yüksek öncelikliler kendi arasında düşük öncelikliler kendi arasında sıraya giriyor.)

When a packet is received at the router the classifier performs a classification and puts the packet in a specific queue based on the classification result.

Packet Schedular

Manages the forwarding of different streams using a set of different streams using a set of queues and timers to schedule the packets to meet their QoS requirements.

Disadvantages of IS

The amount of information increases proportionally (orantılı olarak) with the number of flows.

Since all nodes have to implement (RSVP, admission control, classifier, packet schedular) it places high demand (ihtiyaç) on routers.

Guaranteed service must be provided at each node in the network.

Time varying and location-dependent bw of wireless network is also a problem for IS.

Differential Services (Diff.-Serv.)

Proposed as a response to the scalability (ölçeklendirilebilir) problems in IS concept

Reduces the state of information stored in the network compared to IS architecture.

Based on class identification by using the DS header field, as TOS of IPV4 or traffic class of IPV6, in DS field 6 of 8 bits are used to specify the QoS requirements.

Basic principle of DS is packet forwarding treatment (işlem) (PHB (per hop behaviour))

If handling else is specified BE service will be used.

DS conceptually differs from IS the number of classes is limited within DS due to the limited size of DS field.

Amount of information stored at a network node is proportional to the number of classes rather than to the number of flows.

Differentiated and Integrated Services

Ekstra

How to premium service (Diffserv) and guaranteed service (Intserv) differ in their semantics and implementation?

Semantic Differences:

- The guaranteed service is end-to-end; the premium service is ingress-to-egress (i.e., the premium service is defined on a domain basis)
- The guaranteed service provides both per-flow delay and bandwidth guarantees; the premium service provides only per-flow bandwidth guarantees

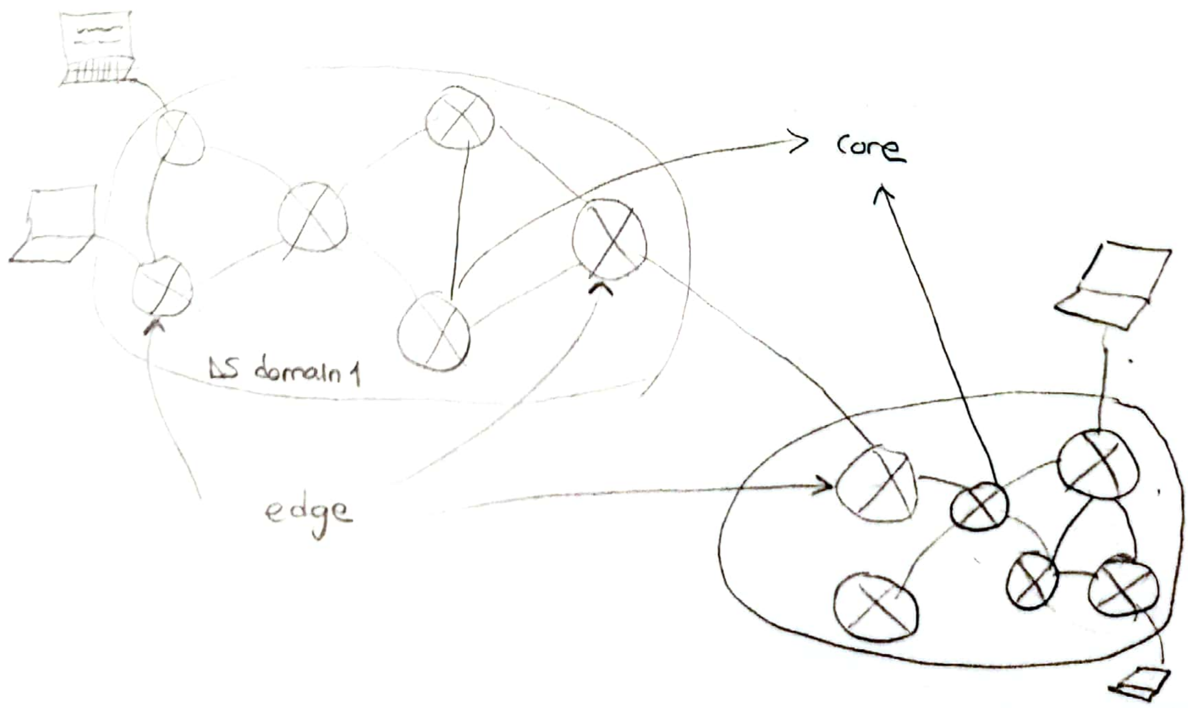
Implementation Differences:

- Premium service (and Diffserv in general) is deployed on a domain basis, and differentiate between edge and core routers.
- With the guaranteed service, every router maintains per-flow state and performs per-flow packet processing and admission control; with the premium service, core routers maintain no per-flow state
- with the guaranteed service, the admission control is totally distributed with the premium service, the admission control is centralized within each domain

DiffServ architecture

Build around the concept of a domain:

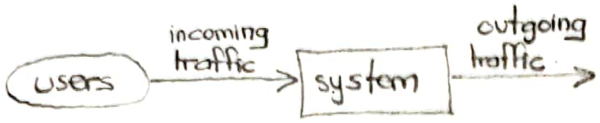
Contiguous region of a network using the same DS configuration



Teletraffic Theory

* Purpose of Teletraffic Theory

Telecommunication system from the traffic point of view



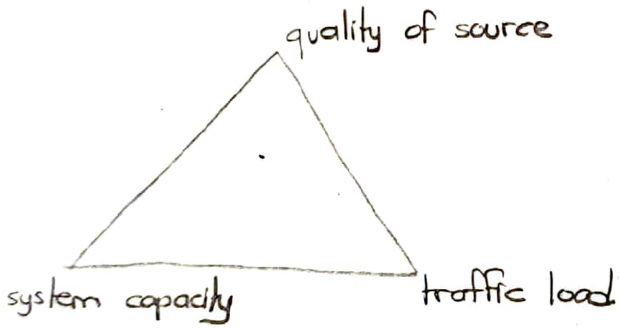
* Ideas

- The system serves the incomming traffic
- The traffic is generated by the users of the system

* General Purpose

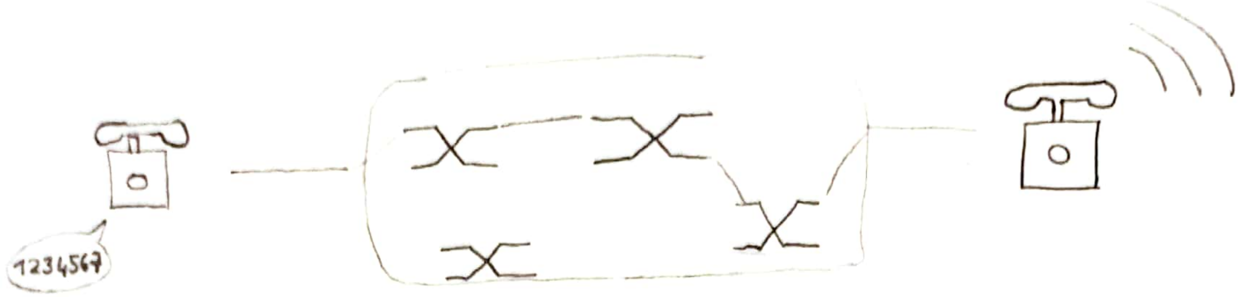
Determine relationship between the following three factors

- quality of source
- traffic load
- system capacity

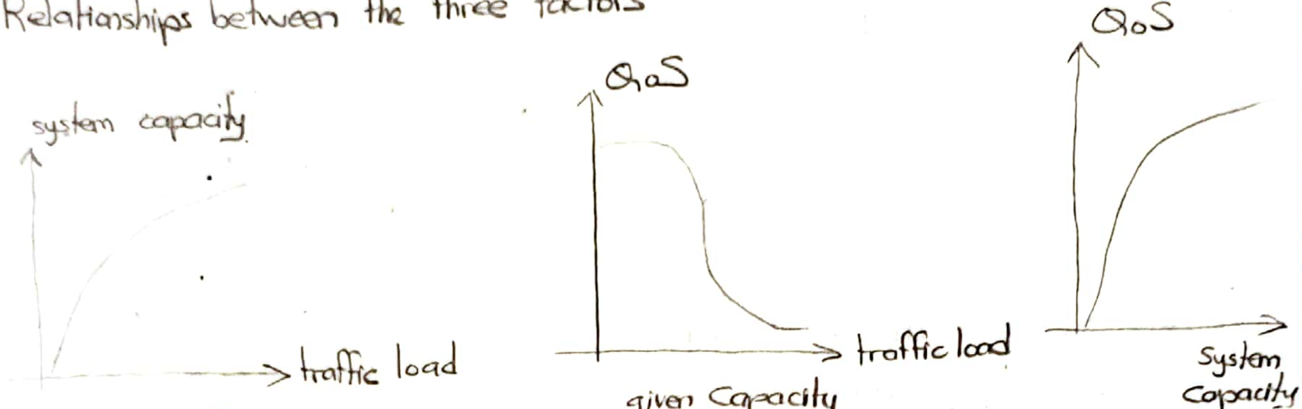


* Telephone traffic

- system = telephone network (for example PSTN)
- traffic = telephone calls by everybody
- quality of service = probability that the connection can be set up, i.e. "the line is not busy."



Relationships between the three factors



given QoS
(sabit olan QoS)

given Capacity
(sabit olan Capacity)

given traffic load
(sabit olan traffic load)

Teletraffic Models

Teletraffic Models are probabilistic → olasılıksal

- Systems themselves are usually deterministic. → rasgele olmayan
  But traffic is typically sthochastic (probabilistic)
- You never know who calls you when
- It follows that the variables in these models are random variables e.g.
  - number of ongoing calls → sürmekte olan
  - number of packets in a buffer
  - Random variable is described by its distribution → dağılım
  - Probability that there are n ongoing calls → sürmekte olan
  - Probability that there are n packets in the buffer
- Stochastic process describes the temporal development of a random variable. → zamanla değişen

Teletraffic Theory

Practical Goals,

- Dimensioning
- Optimisation
- Performance Analysis

Network Management and Control

- Efficient operating (Kaynakları etkili kullanmak)
- Fault Recovery (hata kurtarma)
- Traffic Management
- Routing (iletme)
- Accounting (hesaplama)

Telecommunication Network

A simple model of a telecommunication network consist of

- nodes
- terminals
- networks nodes
- links between nodes

Access Network

- Connects the terminals to the network nodes

Trunk Network

- connects the network nodes to each other.

Switching Modes

* Circuit Switching (only connection oriented)
  - Telephone networks
  - Mobile telephone networks GSM
* Packet Switching
  - Data Networks
  - Two Possibilities

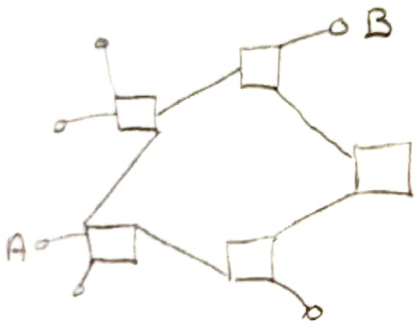
connection oriented — connectionless

→ Each packet is sent with a 'header address' which tells it where its final destination is, so it knows where to go.

* Cell Switching (baz istasyonları arası switching)
  - Fast (connection oriented) packet switching with fixed length packets (called cells). e.g.: ATM (16 byte'lık sabit paketler gönderilir)
  - Integration of different traffic types. (voice, data, video)
  ⇒ multiservice networks.

Circuit Switching

- Connection Oriented
  - connections set up end-to-end before information transfer
  - resources reserved for the whole duration of connection



- Delay occurs before the connection
- No overhead during the information transfer
- Efficient only if connection holding time ≫ connection setup time.

TDM

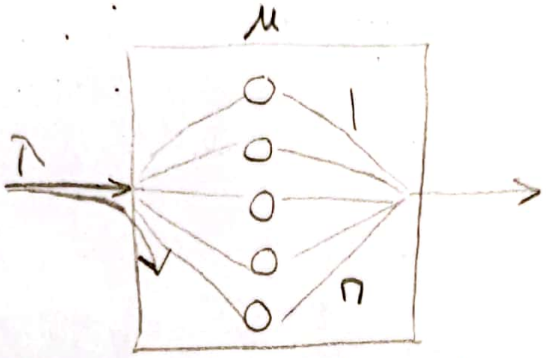
⇒ used in digital cct. sw networks
information conveyed on a link transfer in frames of fixed length.
Location of the time slot within the frame identifies the connection.

Communication via (aracılığıyla) circuit switching has three phases (aşama):
* circuit establishment (link by link)
  - routing & resource allocation (FDM or TDM)
* data transfer
* circuit disconnect
  - deallocate the dedicated resources

Classical Modell For Telephone Traffic

Erlang modelled this as a loss system with n servers.

- costumer = (telephone) call
  - λ = call arrival rate (arama yoğunluğu)
- service time = (call) holding time (bekletme, tutma süresi)
  - h = average holding time
- server = channel on the link
  - n = number of parallel channels on the link

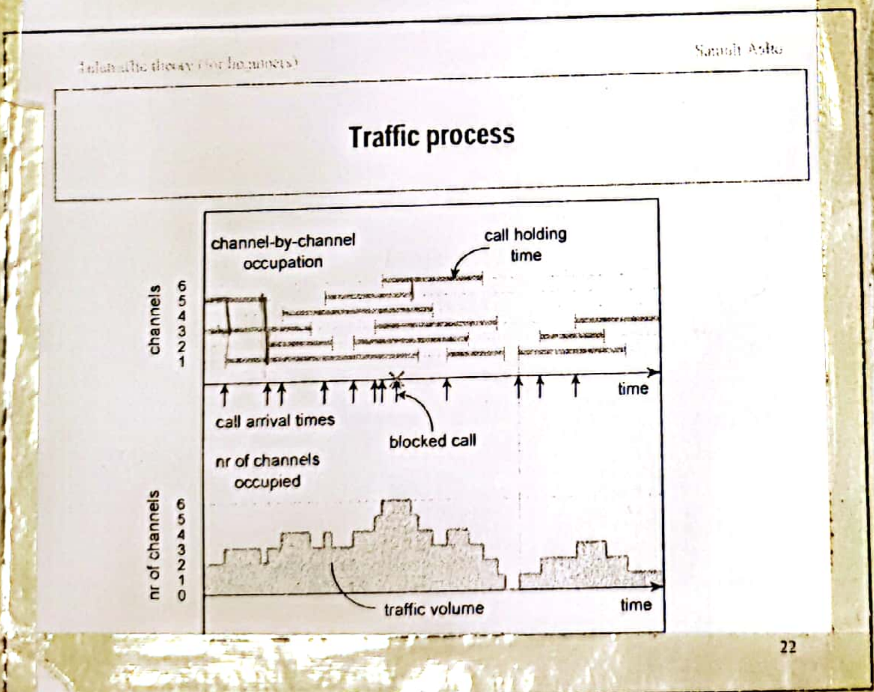


$a \Rightarrow$ traffic intensity

$a = \lambda . h$ erlang

Note that the traffic intensity is a dimensionless quantity the unit of traffic intensity a is called erlang.

Traffic Process



Example = On the average there are 1800 new calls in an hour, and the mean holding time is 3 mins.

$a = (1800 \times 3)/60 = 90$ erlang

↳ dakika cinsinden yazmak için

If mean holding time increases to 10 mins.

$a = (1800 \times 10)/60 = 300$ erlang

90 daha iyi 300'den

Blocking

* In a loss system some calls are lost (kayıpların yaşandığı bir sistemde)
• a call is lost if all n channels are occupied (meşgul) when the call arrives refers to this event. (Hat meşgul olduğunda ekstra biri aradığında konuşamayacak)

* There are (at least) two different types of blocking quantities

• Call blocking B_c = Probability that an arriving call finds all n channels occupied = the fraction of calls that are lost.
↳ (Çağrı yapıldığında hiçbir kanalın boş olmadığı durum.)

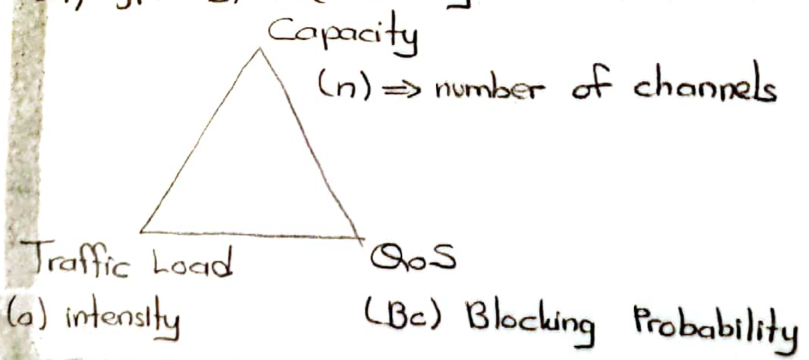
• Time blocking B_t = probability that all n channels are occupied at an arbitrary time = the fraction of time that all n channels are occupied.
↳ (Belli bir zaman veriliyor ve o zamanda kanalların hepsinin dolu olması)

* The two blocking quantities are not necessarily equal. (Example of your own: mobile)
If calls arrive according to a poisson process, then $B_c = B_t$

↳ Herhangi bir zaman aralığında boş kanal bulma olasılığı ile belirli zamanlar arasında boş kanal bulma olasılığı eşit olmaz.

↳ Tüm zamanlara göre bir dağılım verildiyse eğer eşit olur.

Call blocking is a better measure for the quality of service experienced by the subscribers but, typically, time blocking is easier to calculate.



Erlangs Blocking Formula (Quality of Service Hesaplama yapmış oluyoruz.)

If we assume a loss system, that is

- calls arrive according to a poisson process (with rate λ)
- calls holding times are independently and identically distributed according to any distribution with mean h.

Then the quantative relation between the three factors is given by the Erlangs blocking formula.

$$\boxed{B_c = Erl(\underbrace{n}_{\text{kanal sayısı}},\underbrace{a}_{\lambda . h}) = \frac{\frac{a^n}{n!}}{\sum_{i=0}^{n} \frac{a^i}{i!}}}$$

This formula is called as;

- Erlangs formula
- Erlangs B-Formula
- Erlangs loss Formula
- // First Formula

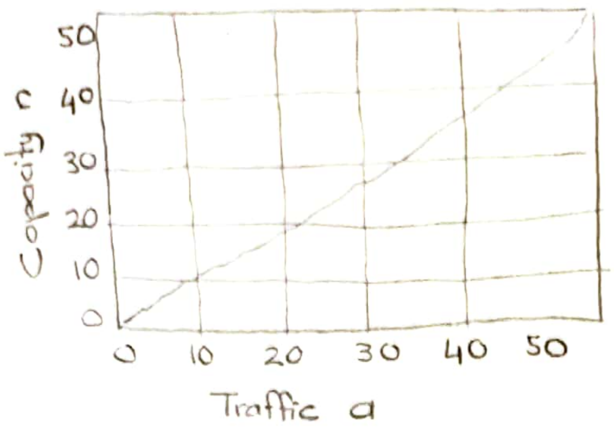
Example = Assume that there are $n=4$ channels on a link and the offered traffic is $a = 2.0$ erlang. Then the call blocking B_c is

$$B_c = Erl(4,2) = \frac{\frac{2^4}{4!}}{1 + 2 + \frac{2^2}{2!} + \frac{2^3}{3!} + \frac{2^4}{4!}} = \frac{2}{21} \approx 9{,}5\ \%$$

$$B_c = Erl(6,2) = \frac{\frac{2^6}{6!}}{1+2+\frac{2^2}{2!}+\frac{2^3}{3!}+\frac{2^4}{4!}+\frac{2^5}{5!}+\frac{2^6}{6!}} = 1{,}2\ \%$$

boş kanal bulamama oranı

Required Capacity V.S. Traffic

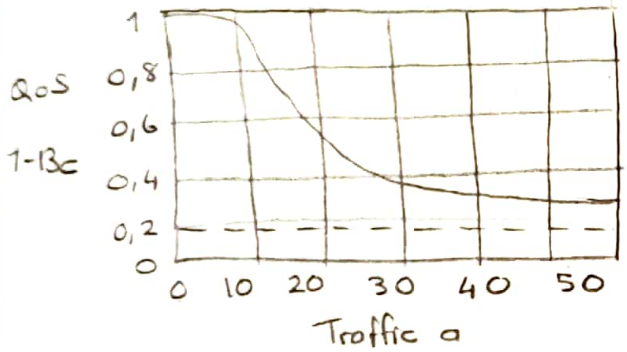


Given that $B_c < 20\ \%$

$n(a) = \min\left(N = 12, \ldots \mid Erl(N, a) < 0{,}2\right)$

Required QoS V.S. Traffic

$\underbrace{1 - B_c(a)}_{\text{Blocke olmama olasılığı}} = 1 - Erl(10, a)$



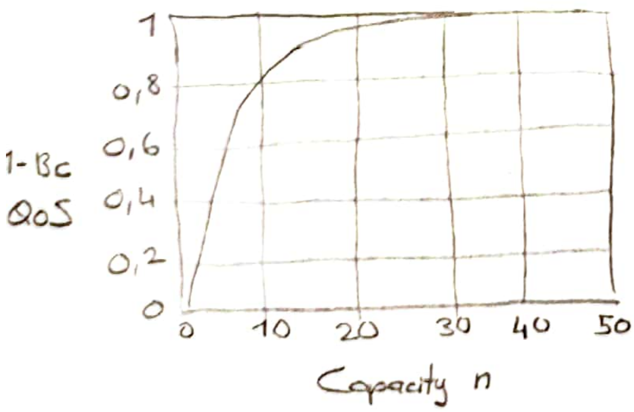
Given that $n = 10$ channels

Traffik arttıkça, QoS düşücek.
(Capacity sabit düşünülüyor)

Traffik yoğunluğunun çok olmaması bizim için iyi degil.
Yüksek olması ve servis kalitesinin yüksek olması iyi. Burda yüksek kapasiteyi gerektirir.

Required Quality of Service VS. Capacity

$1 - B_c(n) = 1 - Erl(n, 10.0)$



(x ekseni arttıkça y ekseninin gözlemliyoruz.)

Given that $a = 10.0$ erlang

Kalitenin sağlanma garantisi
=
Blocke olmama olasılığı

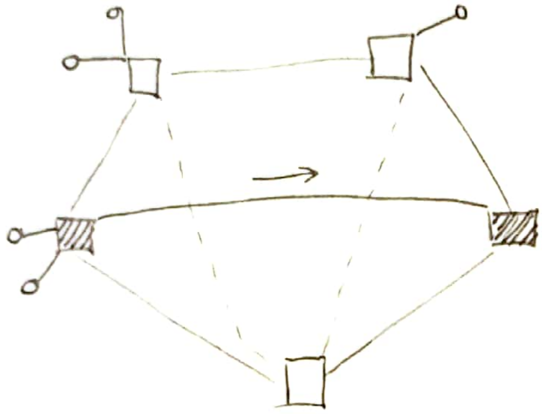
Kanal Sayısı
=
Kapasite

CamScanner ile tarandı

Classical Model For Data Traffic

Queuing models are suitable for describing (packet-switched) data networks

- pioneering work made by ARPANET researchers in 60's and 70's consider a link between two packet routers.
- traffic consist of data packets transmitted on the link

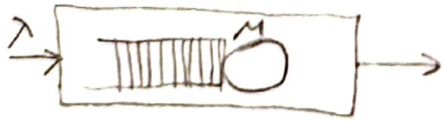


intranet
↓
arpanet
↓
internet

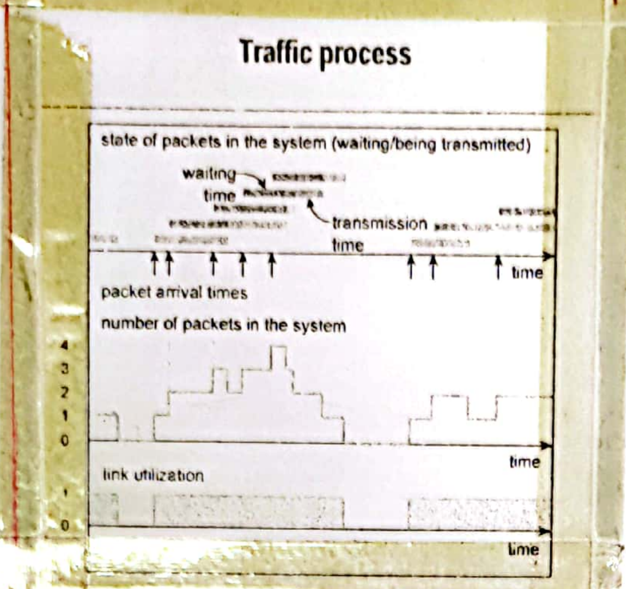
This can be modelled as a waiting system with a single server and an infinite buffer

- costumer = packet
  - λ = packet arrival rate (saniyede çıkan paket)
  - L = average packet length (data units) (yolda nekadar kaldığı)

  bu ikilisinin çarpımı kadar buffer'ı doldurmalı.
- server = link, waiting places = buffer
  - R = Link speed (data units per time unit)
- service time = packet transmission time
  - $1/\mu = L/R$ = average packet transmission time



6 kanalımız var ve 6 kullanıcı bu yolları kullandığı zaman, 7. bir kullanıcı geldiği zaman, bu yollardan herhangi birini kullanamaz. Boşalmasını beklemeli! Yada aynı kanalı kullanması için CDMA gerekli.



Traffic Load

in packet - switched data networks

Traffic ⟷ Packets

– the amount of traffic is described by traffic load ρ.

– By definition, traffic load ρ is the quotient between the arrival rate λ and the service rate $\mu = R/L$

$$\boxed{\rho = \frac{\lambda}{\mu} = \frac{\lambda \cdot L}{R}}$$

– Note that the traffic load is dimensionless quality

– It can also be interpreted (yorumlanmış) as the probability that the server is busy. So it tells the utilization factor of the server.

(Packetin gideceği herhangi bir yol bulamama olasılığını bulursak, packetin kaybolma olasılığını da hesaplamış oluruz.)

Example = Consider a link between two packet routers. Assume that,

- on the average, 10 new packets arrive in a second
- the mean packet length is 400 bytes, and
- the link speed is 64 kbps.

It follows that the traffic load is

byte'ı bite çevirdik.

$$\rho = (10 \times 400 \times 8)/64.000 = 0,5 = 50\ \%$$

If the link speed is increased up to 150 Mbps, the load is just

$$\rho = 10 \times 400 \times 8/150.000.000 = 0.0002 = 0.02\ \%$$

- 1 byte = 8 bits
- 1 kbps = 1 kbit/s = 1,000 bits per second
- 1 Mbps = 1 Mbit/s = 1,000,000 bits per second

Teletraffic Analysis

- System capacity
  - R = link speed in kbps
- Traffic load
  - λ = packet arrival rate in packet/s (considered here as a variable)
  - L = average packet length in kbits (assumed here that L = 1 kbit)
- Quality of service (from the users' point of view)
  - P_z = probability that a packet has to wait "too long", i.e., longer than a given reference value z (assumed here that $P_z = 0.1$s)
- If we assume an M/M/1 queueing system, that is
  - packets arrive according to a Poisson process (with rate λ)
  - packet lengths are independent and identically distributed according to exponential distribution with mean L.
- Then the quantitive relation between the three factors is given by the following waiting time formula.

Waiting time formula for an M/M/1 queue

$$P_z = \mathrm{Wait}(R, \lambda; L, z) = \begin{cases} \frac{\lambda L}{R} \exp\left(-\left(\frac{R}{L} - \lambda\right) z\right), & \text{if } \lambda L < R \ (\rho < 1) \\ 1, & \text{if } \lambda L \geq R \ (\rho \geq 1) \end{cases}$$

Note: The system is stable only in the former (önceki) case ($\rho < 1$). Otherwise the queue builds up (artırmak) without limits.

Example = Assume that packets arrive at rate $\lambda = 50$ packet/s and the link speed is $R = 64$ kbps. Then the probability P_z that an arriving packet has to wait too long (i.e., longer than $z = 0.1$s) is

$$P_z = \mathrm{Wait}(64, 50; 1, 0.1) = \frac{50}{64} \exp(-1.4) \approx 19\ \%$$

Note that the system is stable, since

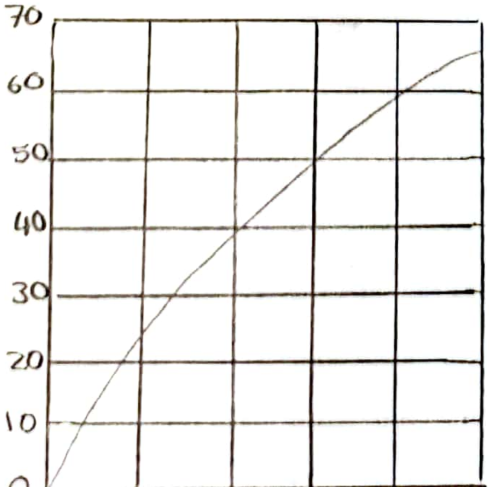
$$\rho = \frac{\lambda . L}{R} = \frac{50}{64} < 1$$

Required Link Speed vs. Arrival Rate

Given the quality of service requirement that $P_z < 20\ \%$, required link speed R depends on arrival rate λ as follows:

$$R(\lambda) = \min\{ r > \lambda . L \mid \mathrm{Wait}(r, \lambda; 1, 0.1) < 0.2 \}$$

link speed R



arrival rate λ

Micromobility

Mobility with in a domain

— Administrative domain (yönetim alanı)

— Campus, building, ISP Network

We opine, → düşüncelerini söylemek, fikir yürütmek

Domain should be big enough to leverage

Micromobility protocols adugs

Micromobility is important because of scalability → ölçeklenebilirlik.

As the no of mobile nodes grow, load on the network increases alarmingly. (networkteki yük çok artmaya ve alarm vermeye başlıyor.)
endişe verici

Processing and signaling load

All this is due to lack of Hierarchy (Hierarcyh olmadığı için bu sıkıntılar oluyor.)

A study has found (at '95) that 69 % of users mobility is local

Why micromobility management?

Mobile

Simple

Scalable

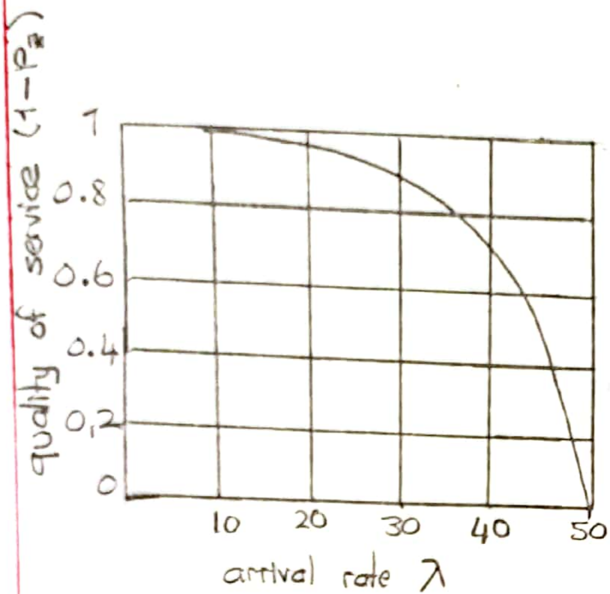
Global Mobility

Lacks support for paging

Required Quality of Service vs. Arrival Rate

Given the link speed $R = 50$ kbps, required quality of service $1-P_2$ depends on arrival rate λ as follows:

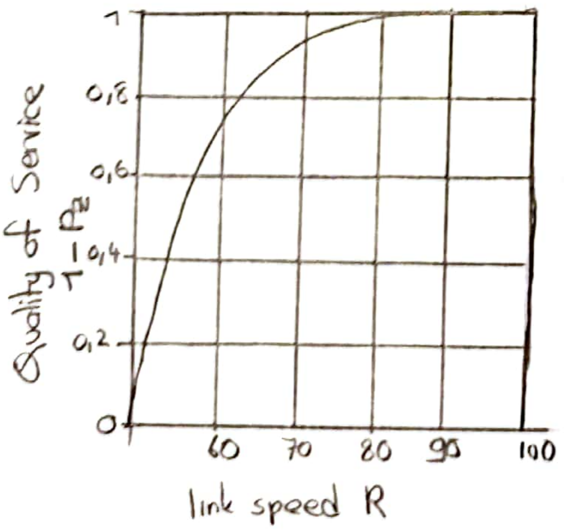
$$1 - P_2(\lambda) = 1 - \text{Wait}(50, \lambda; 1, 0.1)$$



Required Quality of Service vs. Link Speed

Given the arrival rate $\lambda = 50$ packet/s, required quality of service $1-P_2$ depends on link speed R as follows:

$$1 - P_2(R) = 1 - \text{Wait}(R, 50 : 1, 0.1)$$



Characterisation and Classification of IP Traffic

IP traffic is fundamentally differs from the traditional telephony traffic. IP traffic should be characterized and classified to design and IP network. For this, the one must clearly specify the description of the network traffic and performance requirement

IP traffic has two main difference from traditional networks.

1) Resource allocation is dynamic and resource are allocated on a per-packet basis.

2) There is no explicit (belirgin) support for the allocation (tahsis etme) of a specific quantity of network resources.

Characterisation of IP Traffic

nternet traffic Based on IP. Includes many multimedia services that have differences characteristics considering their traffic parameters

- bit rates
- burst iness → ?
- connection duration time
- QoS parameters
- delay
- loss
- throughput → (bütün herşey bunun için hizmet ediyor)
  - ↳ Is the average rate of successful message delivery over a communication channel. This data may be delivered over a physical or logical link or pass through a certain network node.
  - The throughput is usually measured in bits per second (bps or bit/s) and sometimes in data packets per second or data packets per time slot.

bütünleştirmek

Aggregate Internet Traffic

— Best efford service is the dominant one, and means equal bandwidth is shared among all traffic flows.

— Internet service is based on client - server interaction

— Network nodes or terminals segment the information stream into packets, adds headers and trailess for addressing and controlling.

Internet Traffic Components

Internet traffic is usually classified upon the transport protocol (TCP and UDP) or application (web, telnet, FTP, e-mail)

TCP is the dominant protocol in the internet for about 90 % usage and UDP follows it by about 8 %, www amounts for 55 % to 90 % at the TCP traffic others are FTP, SMTP...

UDP is mainly used for real time services, or in combination with the Real Time Protocol (RTP). DNS, TFTP, SNMP

ses ve video gönderiminde kullanılır. throughput ↑ QoS ↑ çok önemli

QoS Classification of IP Traffic

üstünde, üzerinde

The classification of IP traffic will be made upon the QoS demands from different services.

The services existing on the internet for the given QoS are listed as in the table here;

The classification is made according to

- service type (audio, video, data and multimedia)
- distribution of information

dağılım

priority ile çalışıyor

Application	Audio	Video	Data	Real Time	QoS
WWW	—	—	X	2	3
IP telephony	X var	— yok	—	1	1
Multimedia Conference	X	X	X	1	1
Audio Streaming	X	—	—	2	2
Video Streaming	X	X	—	2	2
File Download	—	—	X	3	3
Email	—	—	X	3	3
Multimedia Mail	X	X	X	3	2
E-Commerce	—	—	X	1	1
Service on ?	X	X	X	2	2

1 en iyi
2 orta
3 düşük

Kullanıcının acelesi yok 3
Hemen gitmesi gerekiyorsa 1

Most common applications on todays internet don't have requirements for real time services. Examples include www and e-mail.

These applications use best-efford service, which is the basic service of the current internet.

From the users perspective, one can classfy applications in three main groups,

— interactive applications (e.g. IP telephony)
— distributive services (e.g audio or video streaming)
— services on demand (e.g. e-mail, videos or audio on demand or data transfers)

A global classification of internet app (uygulama) is proposed (önerilen) by two main traffic class.

CLASS A: traffic with QoS support, serviced with higher priority.

CLASS B: traffic without QoS support, serviced with lower priority

Class A is also divider into three subclasses these are

- subclass A1: traffic with higher priority of all
- subclass A2: " " variable bit rate and support for realtime communication (V. Bit Rate) (Değişken Bit Oranı)
- subclass A3: best efford traffic which is defined as class Δ.

Since it is dominant on the internet today, we use traces of TCP traffic for analysis measurements.

A TCP or www trace file is a sequence of rows of data where each row contains data for a single IP packet such as; time when packet arrives at the network node that collects the data, IP address (usually they're masked due to the users)

TCP port numbers at both and nodes information field length.