SatNEx Summer School

QoS for Multimedia Traffic via Satellite

Dr. José Ignacio Moreno Novella

<joseignacio.moreno@uc3m.es>

Summary

- Introduction
- Application Taxonomy and characterization
- Multimedia Traffic Patterns
- QoS support
  - UMTS
  - IETF
- TCP Problems over Satellite

Objective

- To present QoS support for Multimedia Traffic
- We will focus on IP considering Satellite as Access network and implications.
- Review Traffic Patterns and QoS Models
- Introduce new challenges for multiaccess terminals and multimedia signalling protocols.

Definition of QoS

- *Quality of Service (QoS)* refers to the capability of a network to provide better service to selected network traffic over various technologies
- The primary goal of QoS is to provide priority including dedicated bandwidth, controlled jitter and latency (required by some real-time and interactive traffic), and improved loss characteristics. Also important is making sure that providing priority for one or more flows does not make other flows fail.
- On intuitive level, QoS represents quantities like *how fast* can data be transferred, how much does the receiver have to *wait*, how *correct* is the received data likely to be, how much data is likely to be *lost*, etc. ..

QoS Parameters

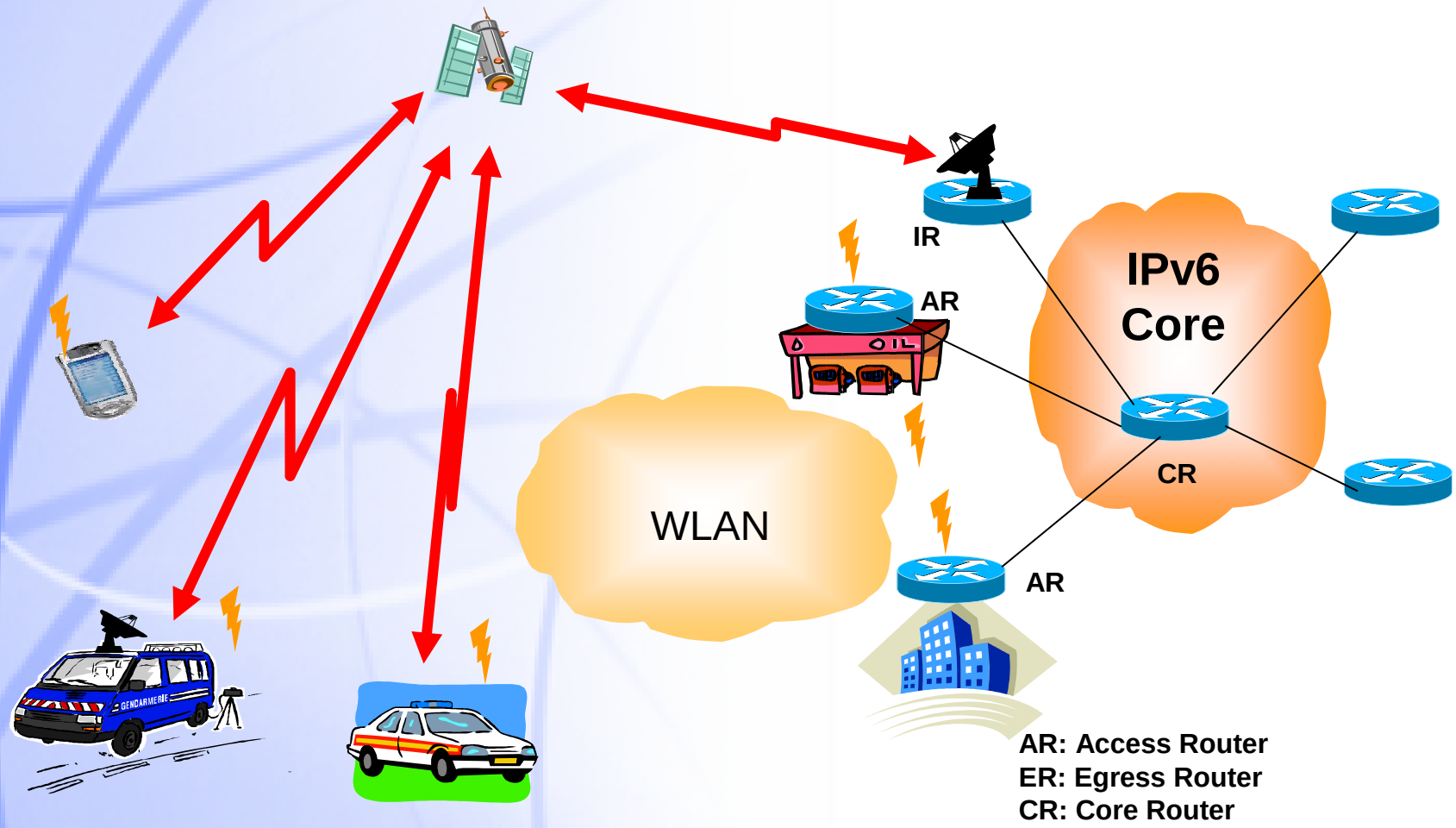
- **Rate**: The Desired Bite Rate (bps) or bandwidth
- **Latency**: Delay encountered by a packet, the sum of transmission delay, processing delays (includes router look-up), queueing delay etc.
- **Jitter**: Variations in Latency
- **Error Rate**: The percentage of packets received in error
- **Loss Rate**: Percentage of packets dropped or lost during end-to-end transmission.

QoS in Computer Networks

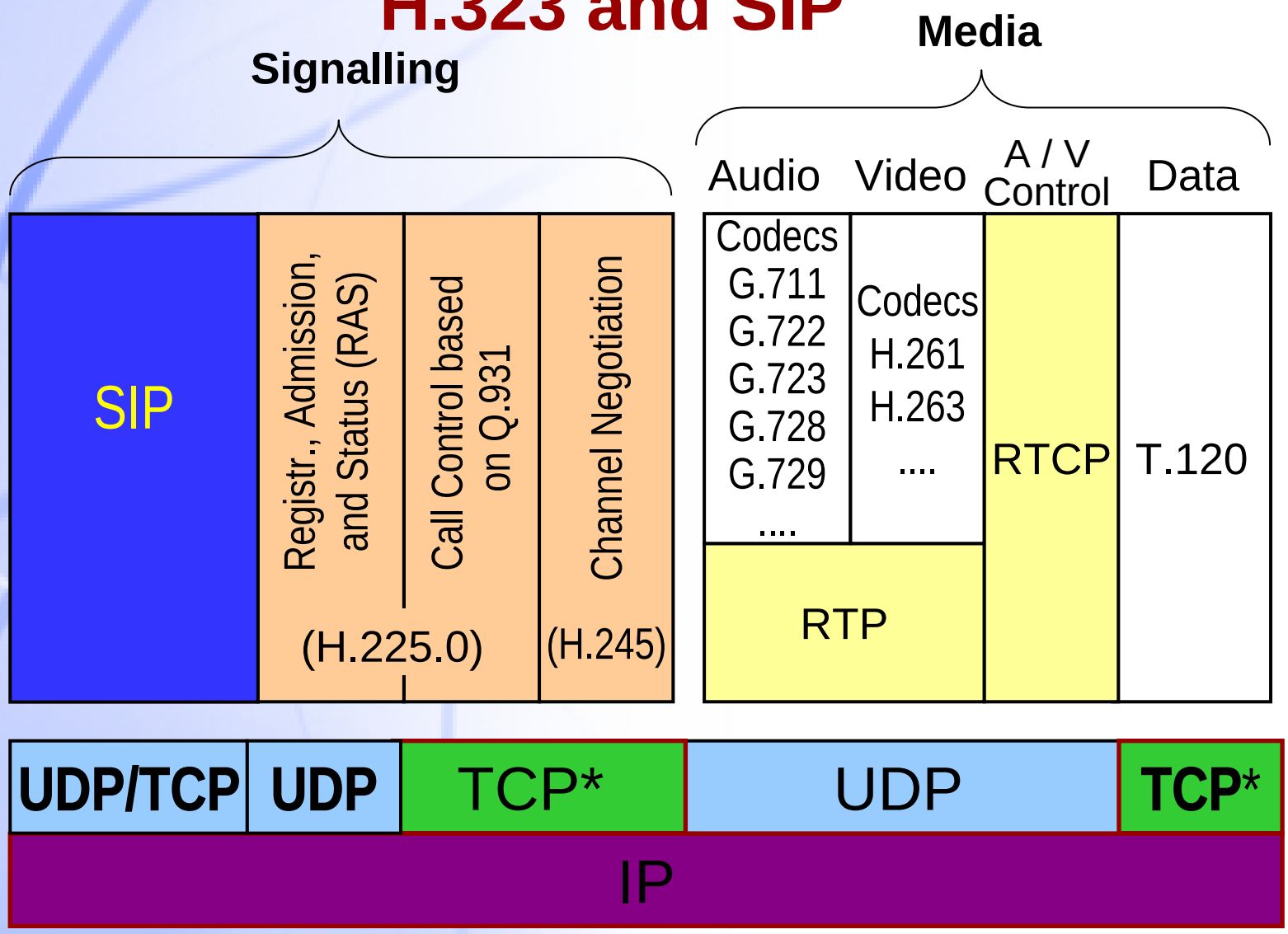
- *"The Holy Grail of computer networking is to design a network that has the flexibility and low cost of the Internet, yet offers the end-to-end quality-of-service guarantees of the telephone network."*

S. Keshav

Scenario

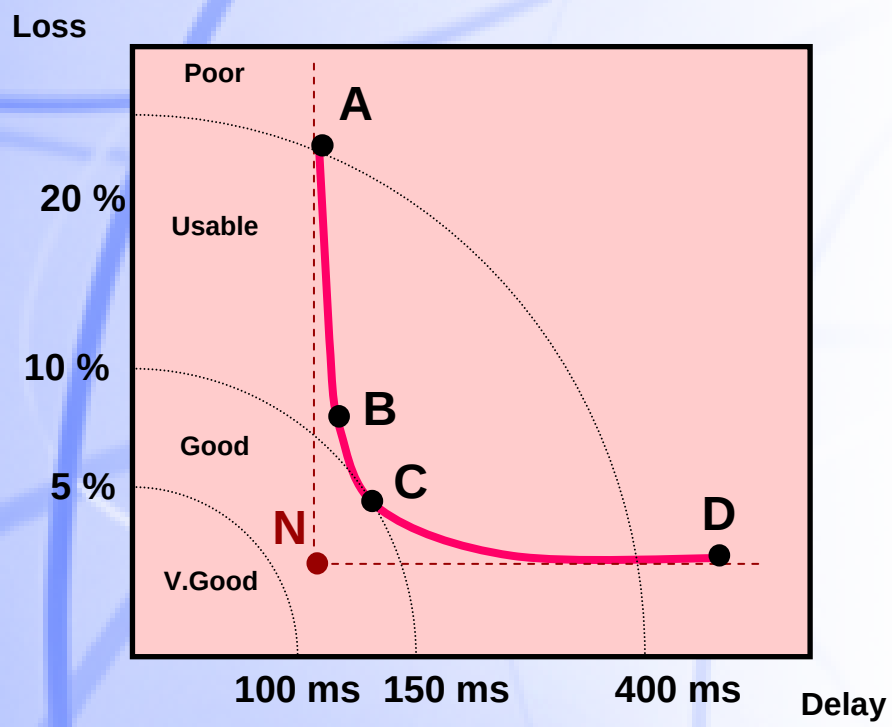

IR
AR
IPv6
Core
CR
WLAN
AR
AR: Access Router
ER: Egress Router
CR: Core Router

Session Management and Multimedia Data for H.323 and SIP



** UDP at H.323 v3*

Buffering Challenge:
Delay and Loss Compromise



ITU-T G.114: Voice one-way Transmision Time

N **Network delay and loss**

B **Working Point**

C **Increase buffer: increase delay reduce loss**

D **Bigger Buffer → Bigger Delay, loss near network loss**

A **Minimize Buffer → Delay near Network Delay, High loss**

Application requirements to the end-users

- Key parameters impacting the user
  - Data Rate
  - Delay, including terminal+network+server, also throughput is affected from end user perspective
  - Delay variation
  - Information loss ratio

- Various applications can be classified into different groups, based on their latency and error tolerance requirements.

Applications Taxonomy

- **Audio**
  - Conversational Voice
  - Voice Messaging
  - Streaming Audio
- **Video**
  - Videophone
  - One-way-video
- **Data**
  - Web-browsing
  - Bulk Data
  - High priority Transaction services (E-commerce)
  - Command/Control
  - Still image
  - Interactive Games
  - Telnet
  - E-mail
  - Instant messaging
  - Background Applications
  - Fax
  - Low Priority Transaction services
  - Email

QoS Audio level based on ITU F.100

A(–1)	sound signal with minimum quality to allow the presence of the speaker to be detected
A(0)	minimum sound quality, sufficient for understanding speech from various speakers; the quality should be at least as good as that obtained with G.711 or G.728 codecs;
A(1)	7 kHz speech quality, at least equivalent to that of a G.722 codec operating at 48 kbit/s;
A(2)	15 kHz broadcast quality sound/speech, at least equivalent to that of a J.41 codec;
A(3)	20 kHz hi-fi or CD quality.

Service	A0	A1	A2	A3
PSTN videotelephone	X			
Mobile videotelephone	X	X		
ISDN videotelephone	X	X		
Desktop videotelephone	X	X		
Audiographic conference	X	X	X	
Videoconference		X	X	
Video surveillance	X	X		
Audio/video distribution			X	X
Multimedia retrieval	X	X	X	X

QoS Video level based on ITU F.100

V(–1)	video signal sufficient to allow movement to be detected;
V(0)	minimum videophone quality, sufficient for showing the head of one person so that his identity and his facial expressions may be recognized; the amount of movement that can be tolerated without degraded temporal performance is very limited;
V(1)	basic videophone quality, sufficient for showing a head and shoulder view of one person while being able to observe the lips movements; the amount of movements is limited to those of a seated person in a normal conversation
V(2)	basic videoconference quality, allowing a small group of three seated persons to be shown simultaneously; the amount of movements is limited to those of a normal discussion;
V(3)	television broadcasting quality (ITU-R Recommendation BT.601)
V(4)	high definition television quality.

Service	V0	V1	V2	V3	V4
PSTN videotelephone	X				
Mobile videotelephone	X	X	X		
ISDN videotelephone		X	X		
Desk-top videotelephone		X	X		
Videoconference		X	X	X	
Video surveillance		X	X	X	
Video distribution				X	X
Video retrieval				X	X

Performance for Audio/Video Applications

- Source: G.1010 -

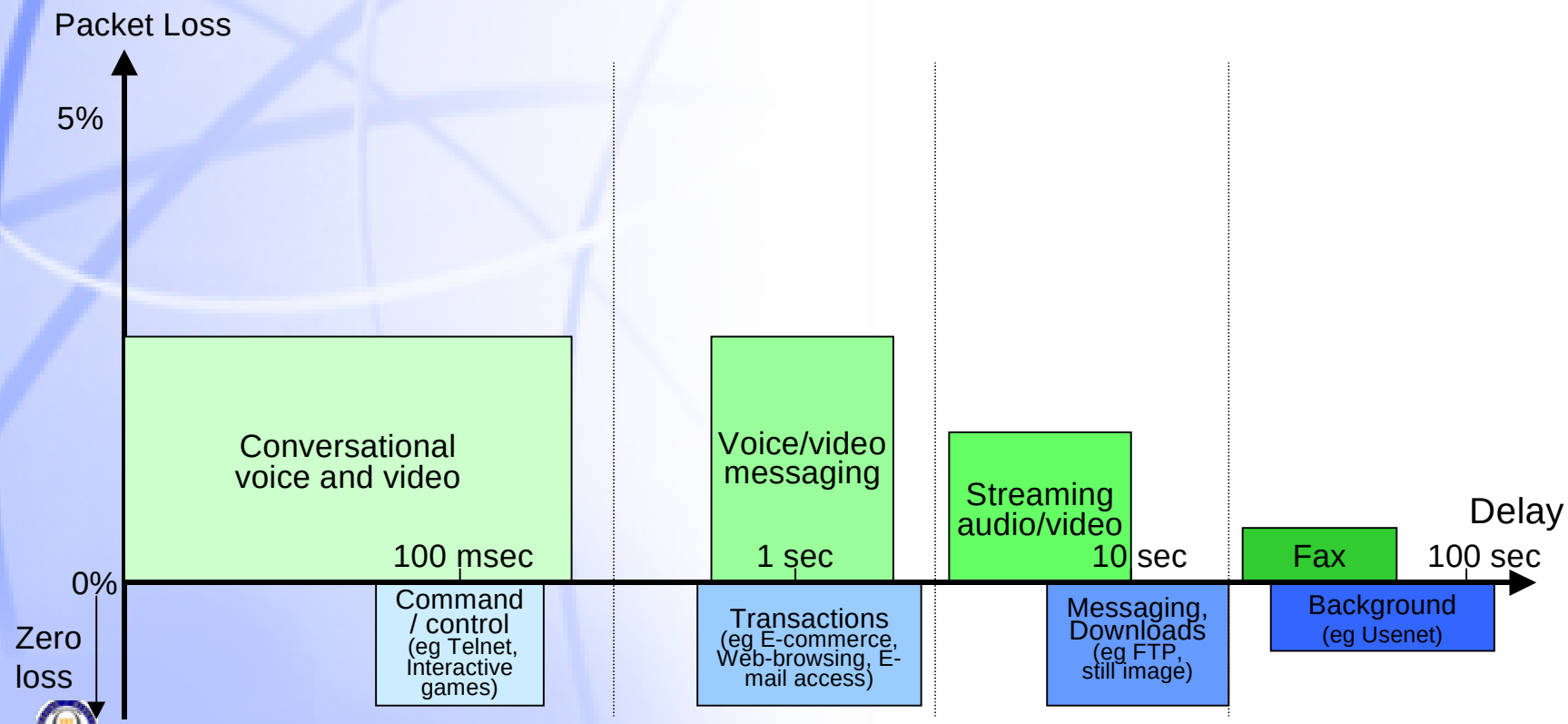
Medium	Application	Degree of symmetry	Typical data rates	Key performance parameters and target values			
				One-way delay	Delay variation	Information loss (Note 2)	Other
Audio	Conversational voice	Two-way	4-64 kbit/s	<150 ms preferred (Note 1) <400 ms limit (Note 1)	< 1 ms	< 3% packet loss ratio (PLR)	
Audio	Voice messaging	Primarily one-way	4-32 kbit/s	< 1 s for playback < 2 s for record	< 1 ms	< 3% PLR	
Audio	High quality streaming audio	Primarily one-way	16-128 kbit/s (Note 3)	< 10 s	<< 1 ms	< 1% PLR	
Video	Videophone	Two-way	16-384 kbit/s	< 150 ms preferred (Note 4) <400 ms limit		< 1% PLR	Lip-synch: < 80 ms
Video	One-way	One-way	16-384 kbit/s	< 10 s		< 1% PLR	

Performance for Data Applications - Source: G.1010 -

Medium	Application	Degree of symmetry	Typical amount of data	Key performance parameters and target values		
				One-way delay (Note)	Delay variation	Information loss
Data	Web-browsing – HTML	Primarily one-way	~10 KB	Preferred < 2 s /page Acceptable < 4 s /page	N.A.	Zero
Data	Bulk data transfer/retrieval	Primarily one-way	10 KB-10 MB	Preferred < 15 s Acceptable < 60 s	N.A.	Zero
Data	Transaction services – high priority e.g. e-commerce, ATM	Two-way	< 10 KB	Preferred < 2 s Acceptable < 4 s	N.A.	Zero
Data	Command/control	Two-way	~ 1 KB	< 250 ms	N.A.	Zero
Data	Still image	One-way	< 100 KB	Preferred < 15 s Acceptable < 60 s	N.A.	Zero
Data	Interactive games	Two-way	< 1 KB	< 200 ms	N.A.	Zero
Data	Telnet	Two-way (asymmetric)	< 1 KB	< 200 ms	N.A.	Zero
Data	E-mail (server access)	Primarily one-way	< 10 KB	Preferred < 2 s Acceptable < 4 s	N.A.	Zero
Data	E-mail (server to server transfer)	Primarily one-way	< 10 KB	Can be several minutes	N.A.	Zero
Data	Fax ("real-time")	Primarily one-way	~ 10 KB	< 30 s/page	N.A.	$<10^{-6}$ BER
Data	Fax (store & forward)	Primarily one-way	~ 10 KB	Can be several minutes	N.A.	$<10^{-6}$ BER
Data	Low priority transactions	Primarily one-way	< 10 KB	< 30 s	N.A.	Zero
Data	Usenet	Primarily one-way	Can be 1 MB or more	Can be several minutes	N.A.	Zero

End User QoS Categories as Recommended by ITU-T G.1010

Class	Interactive	Responsive	Timely	Non-critical
Delay	**<< 1 sec**	**~ 2 sec**	**~ 10 sec**	**>> 10 sec**

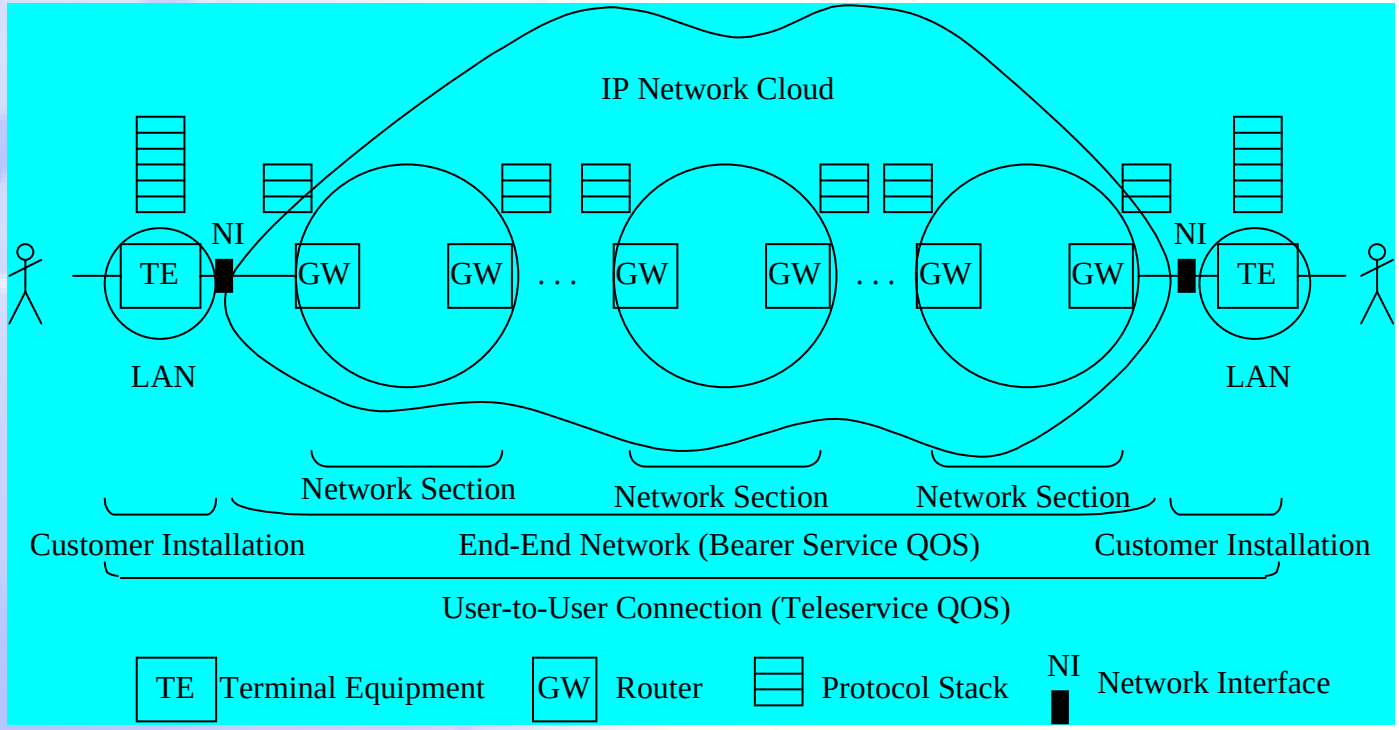


End-user QoS Categories Mapping

Error tolerant	Conversational voice and video	Voice/video messaging	Streaming audio and video	Fax
Error intolerant	Command/control (eg Telnet, interactive games)	Transactions (eg E-commerce, WWW browsing, Email access)	Messaging, Downloads (eg FTP, still image)	Background (eg Email arrival)
	Interactive (delay <<1 sec)	**Responsive** (delay ~2 sec)	**Timely** (delay ~10 sec)	**Non-critical** (delay >>10 sec)

Latency or Packet Transfer Delay

- ITU-T Y.1541 Model: Hypothetical Reference Path for Transfer Delay and Delay Variation Computation
- Path analysis for IPTD & IPDV in an IP network
- End-to-end One Way Transfer Delay = Path Delay + End-point Delay



Error Rate Requirements in Current Standards

- "Network performance objectives for IP-based services": ITU-T Y.1541
  - Parameters that are used in Y.1541 [4] are defined in ITU-T Y.1540
    - IP packet transfer delay (IPTD)
    - IP packet delay variation (IPDV)
    - IP packet loss ratio (IPLR)
    - IP packet error ratio (IPER)
  - Six different classes of traffic based on IPTD and IPDV objectives

Network Performance Parameter	Class 0	Class 1	Class 2	Class 3	Class 4	Class 5
IPTD	100 ms	400 ms	100 ms	400 ms	1 s	U*
IPDV	50 ms	50 ms	U	U	U	U
IPLR	$1x10^{-3}$					U
IPER	$1x10^{-4}$					U

*U: Unspecified

Video over IP Error Rate Requirements

- For real-time video signal, ITU-T SG13 has recommended that IPLR must be at least 10^{-5} [10]
  - Derived based on a BER of 10^{-9} for typical fiber optic network
  - Worst-case assumptions:
    1. Packet size = 1500 bytes
    2. A bit error caused the whole packet to be lost
- User Datagram Protocol (UDP): used for transport of most video streaming applications
- UDP checksum is enabled
  - => A packet may be discarded because of a single bit error
  - UDP does not allow a re-transmission of the lost packet, the effects of losing a complete video packet could result in serious disruption to the video signal
- UDP checksum is normally disabled in most applications
  - Packets with bit errors will be received including the error bits
  - Depending on the location of the error bits, the effects of the lost may be tolerable to the user at the receiving end

Performance Requirements for Real-time Conversational Class - Voice

- **Conversational Voice**
  - **Acceptable maximum FER = 3%**
  - **General limits for one-way end-to-end delay as recommended by G.114:**
    - **0 - 150 ms - preferred range**
    - **150 - 400 ms - acceptable range with increasing degradation**
  - **Requirement is also dependent on the error resilience of speech codec. For AMR speech codec: [9]**
    - **Bit rate: 4.75 - 12.2 kbps**
    - **Required BER:**
      - 10^{-4} for Class 1 bits, 10^{-3} for Class 2 bits, a higher BER class at 10^{-2} might also be feasible.
      - With codec frame length of 20 ms, the one-way end-to-end delay should be less than 100ms.

Performance Requirements for Real-time Conversational Class - Others

- Videophones
  - Delay requirement: similar to conversational voice, with additional requirement on limits of lip-synch
  - Maximum acceptable FER: 1%
- Interactive games
  - One-way delay value of 250 ms proposed in [8]
  - Detail studies on multiplayer network gaming reported [11] the range of acceptable maximum round-trip time: 200 ms - 40 seconds, depending on the type of games, experience level of players
    - Gaming can be : Conversational, Interactive or Interactive/Background
    - Proposed residual BER requirement: 10^{-3}, 10^{-5} / $6x10^{-8}$, or 10^{-5} / $6x10^{-8}$ respectively
    - Proposed maximum one-way delay for the action game: 80ms.
- Two-way control telemetry
  - One-way delay limit: 250 ms proposed in [8]
  - "0" information loss

Performance Requirements for Interactive Class

- Voice messaging:
  - Similar error rate requirement as that of conversational class
  - Delay: up to a few seconds
- Web Browsing:
  - Delay requirement: 2 - 4 seconds/page
  - Recommended to improve to 0.5 second
- High priority transaction services - E-commerce:
  - Delay: 2 - 4 seconds
  - Information loss: “0”
- Email:
  - Server to Server delay: several minutes or hours
  - User to local server delay: 2-4 seconds

Performance Requirements for Streaming Class – Audio/Video

- Audio streaming:
  - mainly an one-way application from server to user (s)
  - Contents of the audio stream: high quality music or broadcasting
  - Packet Loss rate requirement: 1% [13]
  - Delay requirement: one-way delay = 10 seconds
- Video streaming:
  - Similar to audio streaming as described above
  - MPEG-4 video: [9]
    - Bit rate: 24 - 128 kbps or higher
    - End-to-end delay: 150 - 400ms
    - BER: 10^{-6} - 10^{-3} with significant degradation for the latter
- Still image:
  - Similar requirements to Video Streaming
  - Error tolerance: dependent on the encoding and compression formats

Performance Requirements for Streaming Class - Data

- **Bulk Data Transfer (File Transfers):**
  - **May be classified as streaming service with relaxed delay tolerance**
  - **“0” final information loss at the receiving end**
- **Telemetry applications (Monitoring)**
  - **Error rate and delay requirements: similar to that of the bulk data transfer**

Performance Requirements for Background Class

- No stringent requirement on delay for the background class of services
- Requirements for Fax:
  - Delay: 30 seconds
  - BER: less than 10^{-6} [13]
- Low priority transaction services
  - Short message services (SMS)
  - Delivery delay: 30 seconds
  - Email:
    - Server to server delay: wide range with median of several hours

Traffic Patterns

Packet size at IP level in bytes for the different applications.

Application	Direction: Client (C)→Server (S) Peer A→ Peer B	Mean	Max	Min	Average desviation (Desv)	Desv/Mean
Audio stream[1]		380	380	380	0	0
Quake[2]	C→S	82.12	110	73	3.73	0.05
	S→C	161.79	1051	90	48.12	0.3
VoIP[3]	A→B	93	93	93	0	0
	B→A	93	93	93	0	0
Video Stream[4]		1363.93	1364	424	6.09	0
Tetris[5]	C→S	68.63	330	46	36.12	0.53
	S→C	87.72	329	60	60.45	0.68

Notes:
1) By using Robust Audio Tool (RAT) with Linear 16 codec
2) 6 players match all against all
3) By using RAT with GSM codec
4) By using VideoLAN and a movie trailer of about 2 minutes
5) 6 players match

Traffic Patterns

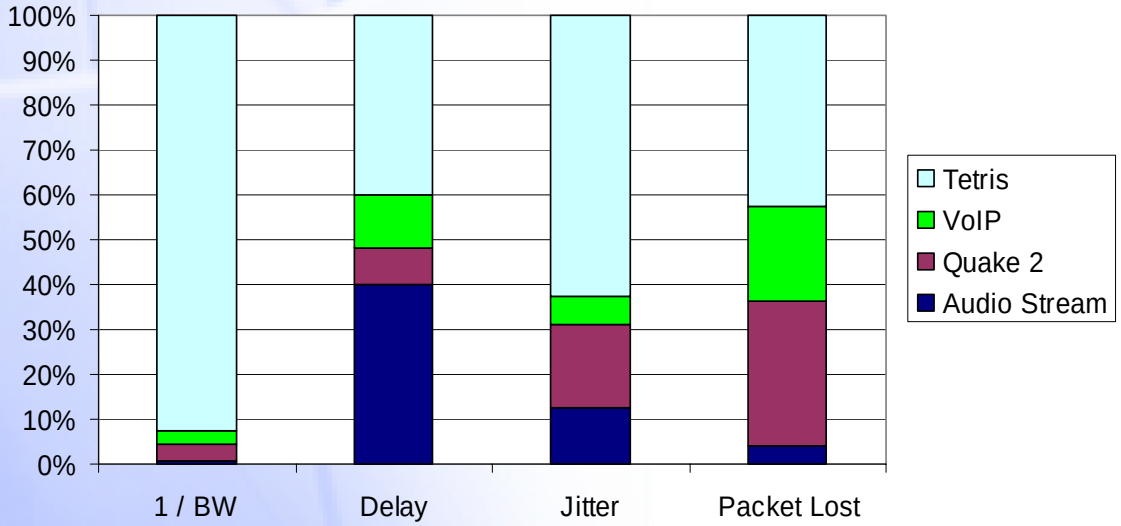
Packets per second generated by the different applications.

Application	Direction: Client (C)→Server (S) Peer A→ Peer B	Mean	Max	Min	Average deviation (Desv)	Desv/Mean
Audio stream[1]		49.33	52	28	2.86	0.06
Quake[2]	C→S	124.35	144	94	6.53	0.05
	S→C	59.95	72	36	4.32	0.07
VoIP[3]	A→B	24.81	52	0	23.52	0.95
	B→A	32.42	52	0	20.67	0.64
Video Stream[4]		172.9	196	18	18.47	0.11
Tetris[5]	C→S	7.64	98	0	11.97	1.57
	S→C	7.63	88	0	11.82	1.55

Notes:
1) By using Robust Audio Tool (RAT) with Linear 16 codec
2) 6 players match all against all
3) By using RAT with GSM codec
4) By using VideoLAN and a movie trailer of about 2 minutes
5) 6 players match

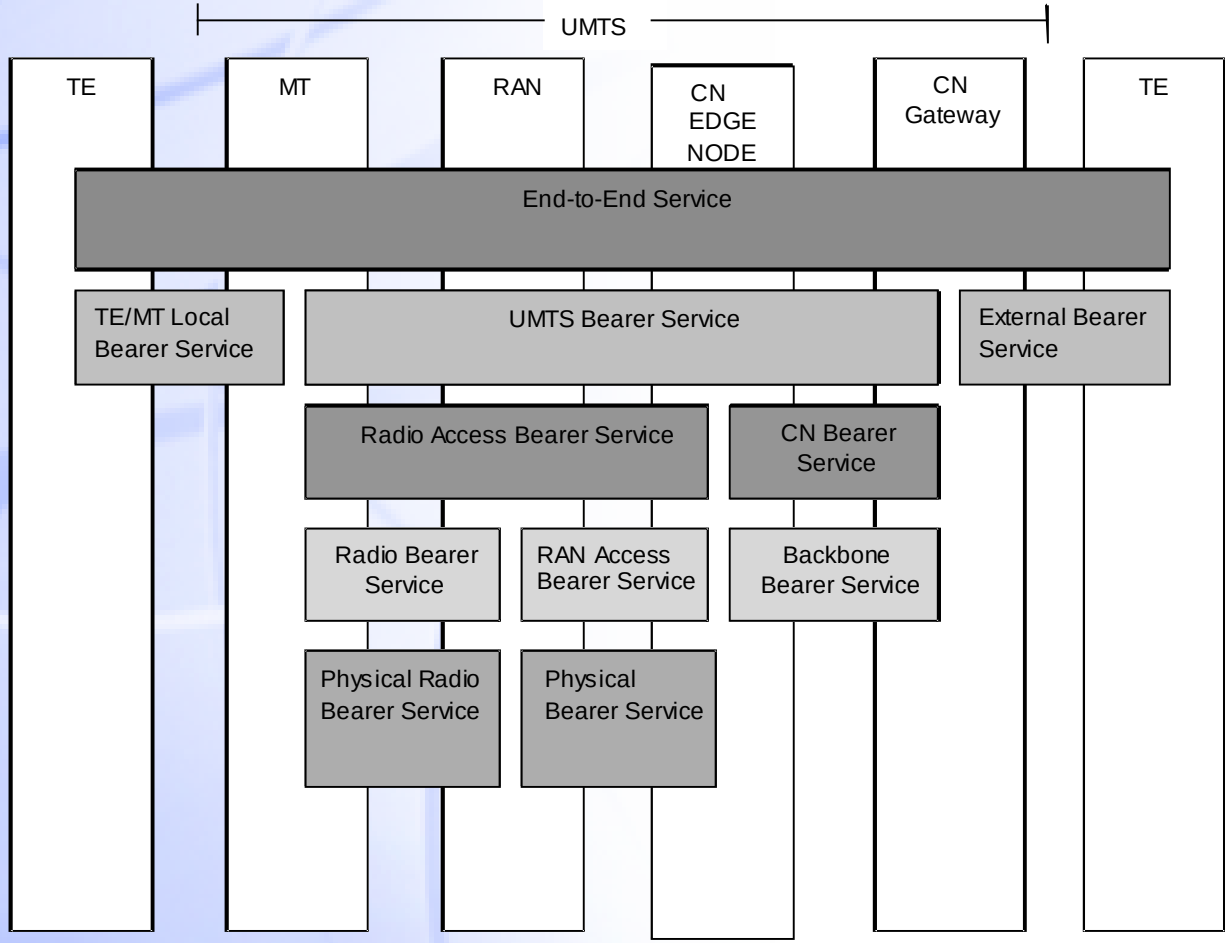
Application Tolerance

Application	Packet Lost	Delay/Direction (ms)	Jitter/Direction (ms)	BW L3 (kbps)
Audio Stream	2 %	>500	100	144
Quake 2	15 %	100	150	24
VoIP	10 %	150	50	30
Tetris	20 %	>500	>500	1



QoS Classes and Support

UMTS QoS Architecture



UMTS QoS classes

- Four different QoS classes are defined in 3GPP TS 23.107:
  - Main difference how delay sensitive the traffic is.

Application Types	QoS Class	Examples	Fundamental Characteristics
Real Time Applications	conversational	Voice , VoIP, Video Conferencing	•Preserve Time relation (variation)between information entities of the stream (Jitter) •Conversational Pattern (stringent and Low delay)
	streaming	Video streaming	•Preserve time relation (variation) between entities of the stream (Jitter)
Traditional Internet Applications	interactive	Web browsing, Data base retrieval, Server access	higher priority than background. Round Trip Time quite relevant. •Request Response Pattern (Low RTT). •Preserve Payload content
	background	Email download, FTP	no delay guarantees •Data transparently transmitted: Preserve Play load content

UMTS QoS bearer attributes

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate	X	X	X	X
Delivery order	X	X	X	X
Maximum SDU size	X	X	X	X
SDU format information	X	X		
SDU error ratio	X	X	X	X
Residual bit error ratio	X	X	X	X
Delivery of erroneous SDUs	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/Retention priority	X	X	X	X
Source statistics descriptor	X	X		
Signalling indication			X	

UMTS Radio Bearer attributes

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate	X	X	X	X
Delivery order	X	X	X	X
Maximum SDU size	X	X	X	X
SDU format information	X	X		
SDU error ratio	X	X	X	X
Residual bit error ratio	X	X	X	X
Delivery of erroneous SDUs	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/ Retention priority	X	X	X	X
Source statistics descriptor	X	X		
Signalling Indication			X	

Ranges of UMTS Bearer Service Attributes

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate (kbps)	<= 16 000 (2)	<= 16 000 (2)	<= 16 000 - overhead (2) (3)	<= 16 000 - overhead (2) (3)
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size (octets)	<=1 500 or 1 502 (4)	<=1 500 or 1 502 (4)	<=1 500 or 1 502 (4)	<=1 500 or 1 502 (4)
SDU format information	(5)	(5)		
Delivery of erroneous SDUs	Yes/No/- (6)	Yes/No/- (6)	Yes/No/- (6)	Yes/No/- (6)
Residual BER	$5*10^{-2}$, 10^{-2}, $5*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6}	$5*10^{-2}$, 10^{-2}, $5*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6}	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (7)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (7)
SDU error ratio	10^{-2}, $7*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}	10^{-1}, 10^{-2}, $7*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}	10^{-3}, 10^{-4}, 10^{-6}	10^{-3}, 10^{-4}, 10^{-6}
Transfer delay (ms)	100 – maximum value	300 (8) – maximum value		
Guaranteed bit rate (kbps)	<= 16 000 (2)	<= 16 000 (2)		
Traffic handling priority			1,2,3 (9)	
Allocation/Retention priority	1,2,3	1,2,3	1,2,3	1,2,3
Source statistic descriptor	Speech/unknown	Speech/unknown		
Signalling Indication			Yes/No (9)	

Ranges for RAB Service Attributes

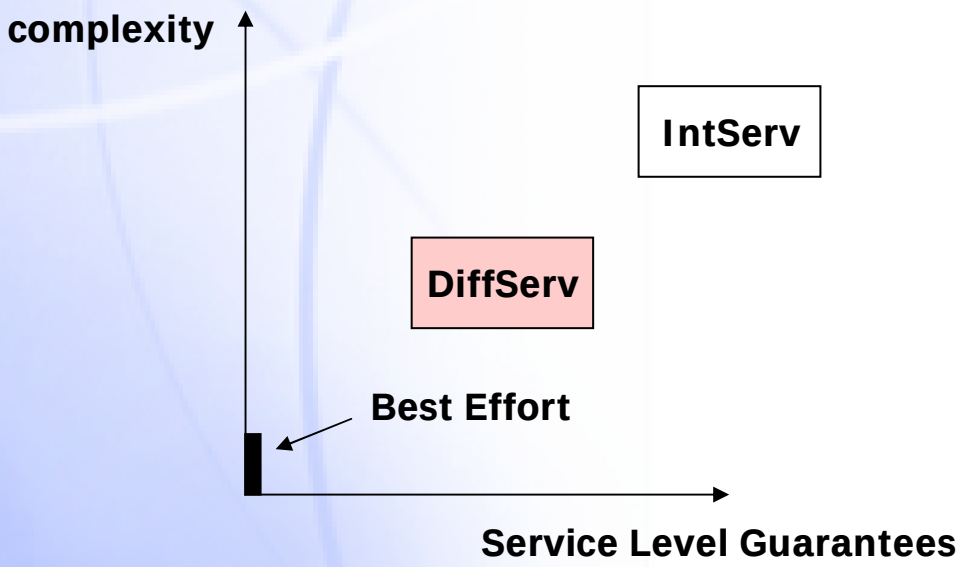
Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate (kbps)	<= 16 000 (2) (7)	<= 16 000 (2) (7)	<= 16 000 - overhead (2) (3) (7)	<= 16 000 - overhead (2) (3) (7)
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size (octets)	<=1 500 or 1 502 (4)	<=1 500 or 1 502 (4)	<=1 500 or 1 502 (4)	<=1 500 or 1 502 (4)
SDU format information (1)	(5)	(5)		
Delivery of erroneous SDUs	Yes/No/-	Yes/No/-	Yes/No/-	Yes/No/-
Residual BER	$5*10^{-2}$, 10^{-2}, $5*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6}	$5*10^{-2}$, 10^{-2}, $5*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6}	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (6)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (6)
SDU error ratio	10^{-2}, $7*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}	10^{-1}, 10^{-2}, $7*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}	10^{-3}, 10^{-4}, 10^{-6}	10^{-3}, 10^{-4}, 10^{-6}
Transfer delay (ms)	80 – maximum value	250 – maximum value		
Guaranteed bit rate (kbps)	<= 16 000 (2) (7)	<= 16 000 (2) (7)		
Traffic handling priority			1,2,3	
Allocation/Retention priority (1)	1,2, ..., 15	1,2, ..., 15	1,2, ..., 15	1,2, ..., 15
Source statistic descriptor	Speech/unknown	Speech/unknown		
Signalling Indication			Yes/No	

IP QoS Models

- Over Provisioning, best effort over big pipes
  - No change in the network
  - Provider control the usability of the network and increase BW accordingly
- Reservation of Resources in advance:
  - Integrated Services (IntServ, RFC)
    - Application should "describe" resources to be sent to the network
    - by using a reservation protocol (RSVP)
    - Routers should negotiate the request resources and guarantee.
    - Information is managed per flow
- Priorization of aggregated flows
  - Differentiated Services (DiffServ, RFC2475, updated by RFC3260)
    - Par-Class QoS guaranties
    - Diferentiation based on "Per-Hop-Behaviour" (PHB)

QoS Granularity

- IntServ: Per-flow QoS
  - QoS guaranties are specified and enforced for individual end-to-end flows.
- DiffServ: Per-Class QoS
  - QoS guaranties are specified and enforced for groups of flows with similar QoS requirements.



Integrated Service (IntServ)

- IntServ framework was developed within IETF to provide individualized QoS guarantees to individual sessions.
- provides services on a *per flow basis* where a flow is a packet stream with common source address, destination address and port numbers.
- IntServ routers must maintain per flow state information.

IntServ

- two key IntServ features:
  - Reserved Resources
    - the router must know the amount of its resources currently reserved for on-going sessions.
    - standard resources: link capacity, router buffers
  - Call Setup
    - A flow requiring QoS guarantees must be able to reserve sufficient resources at each router on path to ensure QoS requirements are met.

Call Setup details

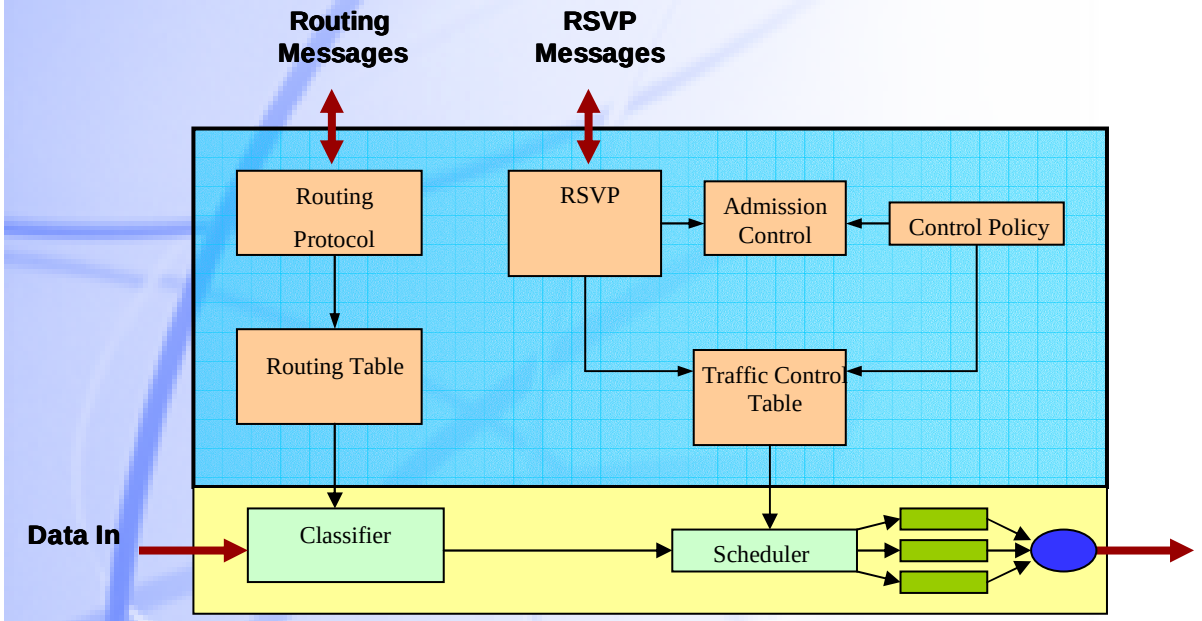
- Call Setup *{also referred to* ***call admission****}*
  - requires participation of each router on the path.
- steps in call setup process
  - Traffic characterization and specification of QoS
    - **Rspec** (R for reserved): defines the specific QoS being requested by a connection.
    - **Tspec** (T for traffic):characterizes the traffic the sender will be sending into the network or the traffic that the receiver will be receiving from the network.

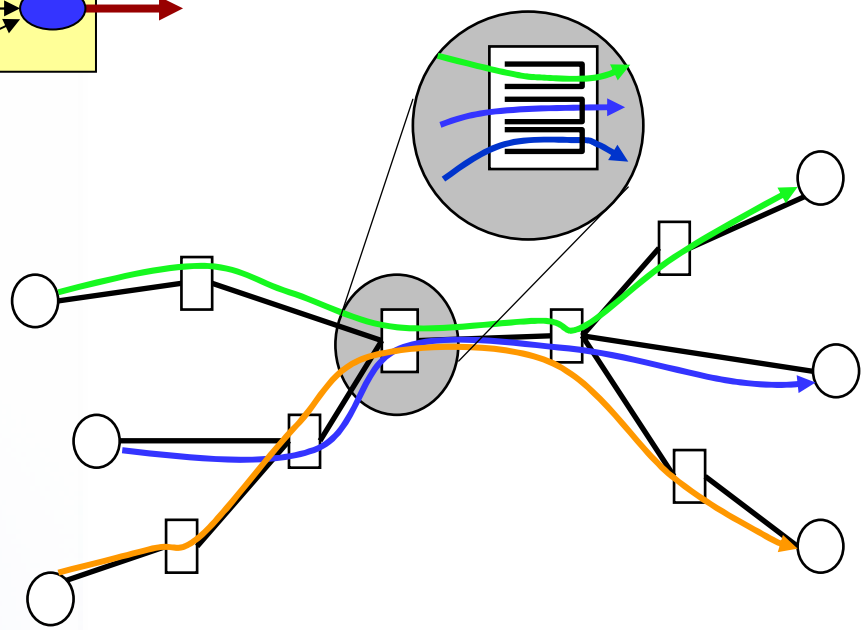
Call Setup details

- **Signaling for call setup**
  - **A session's Tspec and Rspec must be carried to the routers where resources will be reserved.**
  - **RSVP is the signaling protocol of choice.**

- **Per-element call admission**
  - **Once a router receives Rspec and Tspec for a session, it decides whether or not to admit the call.**

IntServ Architecture





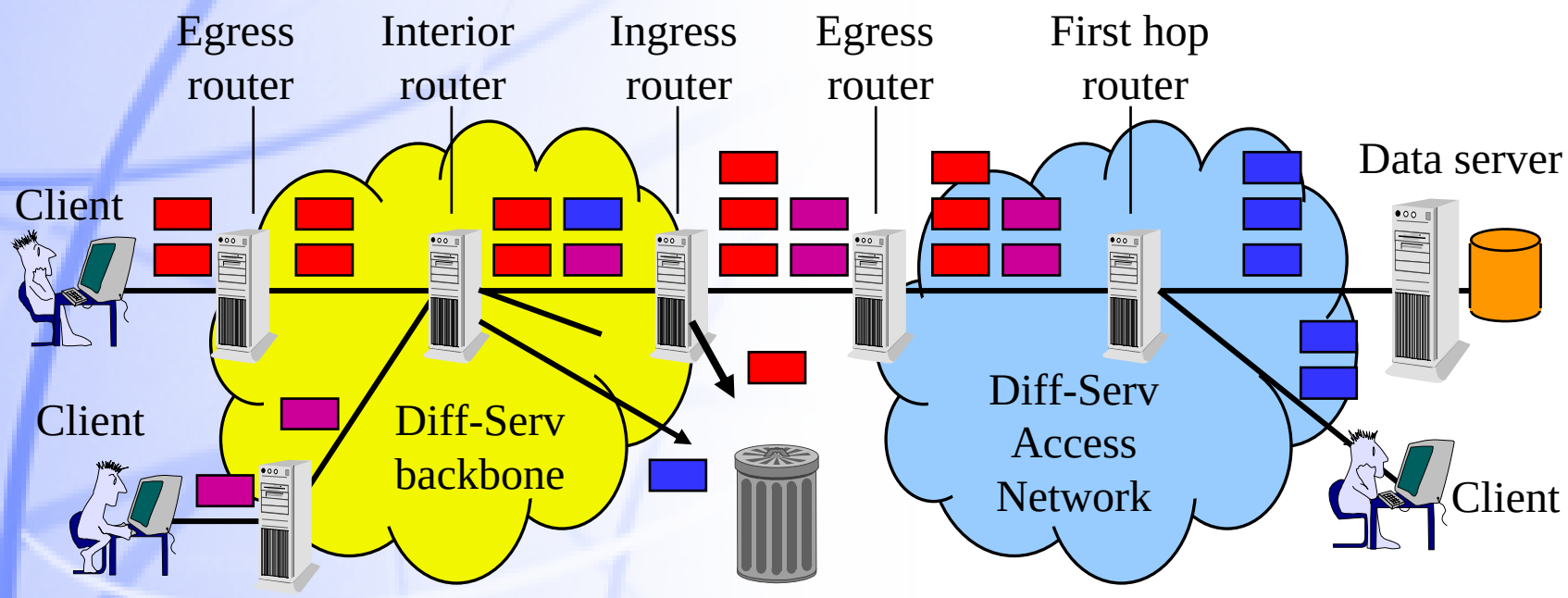
IntServ traffic classes

- **Best Effort Service**
  - No QoS guarantiees
- **Controlled Load service [RFC 2211]**
  - Similar to BE in an unloaded situations
  - No upper bound on queuing delay; high percentage of packets experience delays close to the minimum
  - Almost no queuing lost.
  - This is fine when the network is lightly loaded, but the service degrades quickly as network load increases.
- **Guaranteed Service [RFC2212]**
  - Assured Bandwidth
  - Upper bound on queuing delay
  - No queuing losses
  - A source's traffic characterization is given by a leaky bucket

IntServ Challenges

- Each network elements must be IntServ-enabled
- Each router must maintain per-flow state information and must perform per-flow classification
  - Scalability problems
- Signalling overhead
- Inflexible service model
- Security/Authenticartion control: who can make reservations ?

Principle Idea of Differentiated Service



- **Different kind of marking codes (e.g. red, pink and green for Best-Effort).**
- **Different ways to treat the flows (e.g. red flows will be restricted if it is not compliant / pink flows will be re-marked).**
- **Different kind of routers (first hop, ingress, egress and interior routers).**
- *Differentiated Services are only uni-directional!*

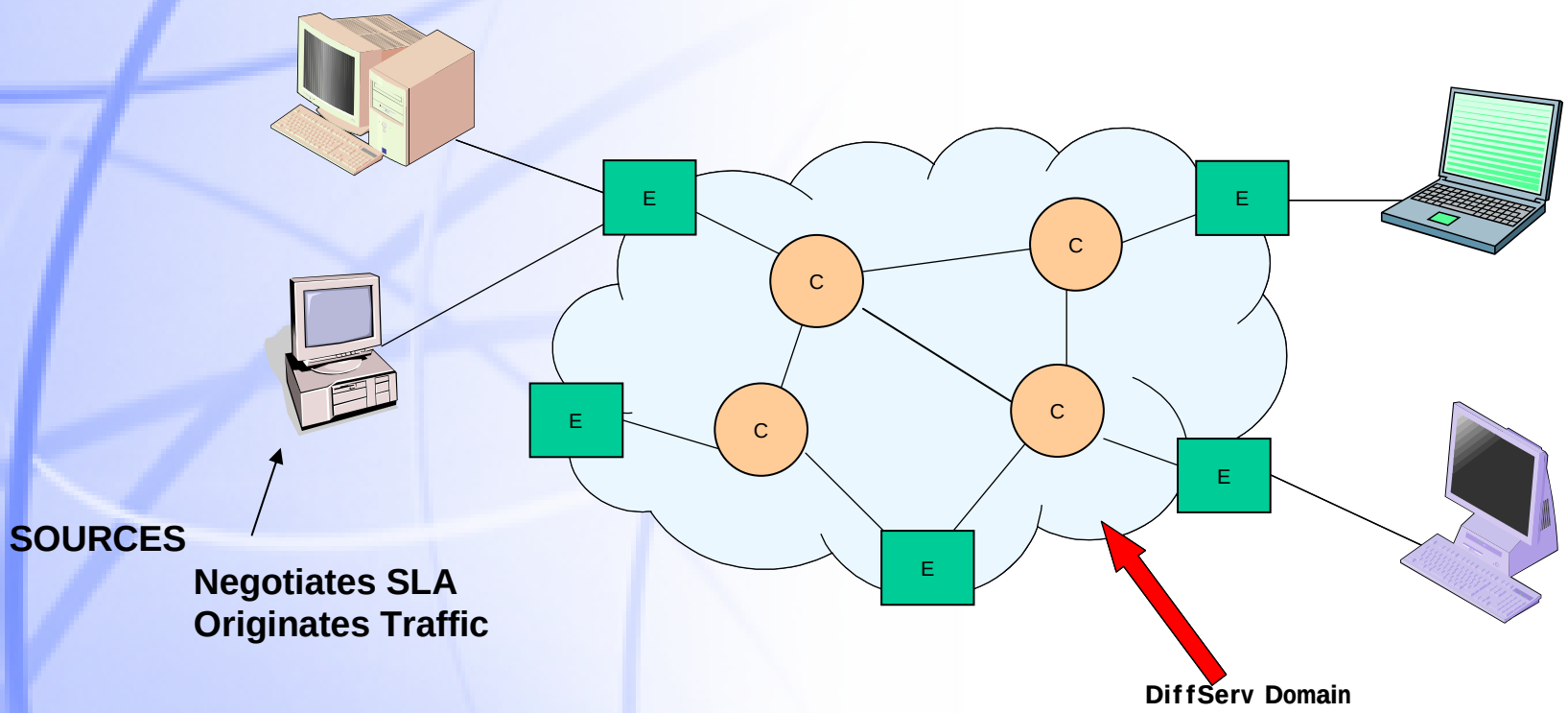
DiffServ

- Provide different quality of service levels without the complexity and overhead of IntServ
- Support for large number of flows
- Offer a limit number of classes for aggregate traffic
- Routers are not aware of individual flows but on traffic aggregates.
- Keep a few traffic classes in the network, based on marking of the packets
- No signalling between routers
- Complexity at the borders of the network
  - Traffic enforcement are done only at the network boundaries.

Diffserv: main concepts

- Per-hop behaviuor (PHB)
  - Forwarding bahaviour applied at a node to a traffic aggregate
- Service provide by thenetwork
  - Composition of PHBs
- DSCP (DiffServ Code Point)
- Service Level Agreement (SLA)
  - Contract between user and a domain or between two domains
- Traffic Conditioning
  - Control Functions performed at network boundaries to enforce SLAs

DiffServ Model

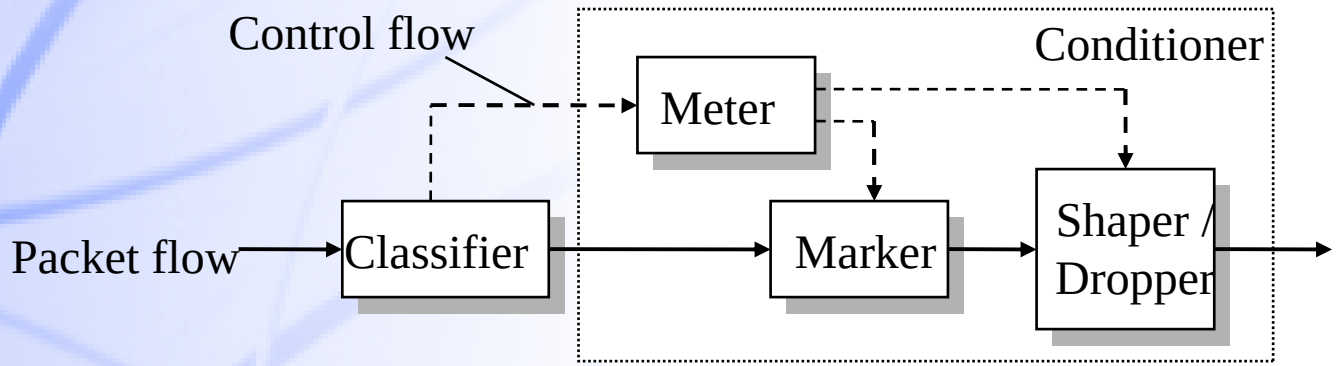


E Edge router

C Core router

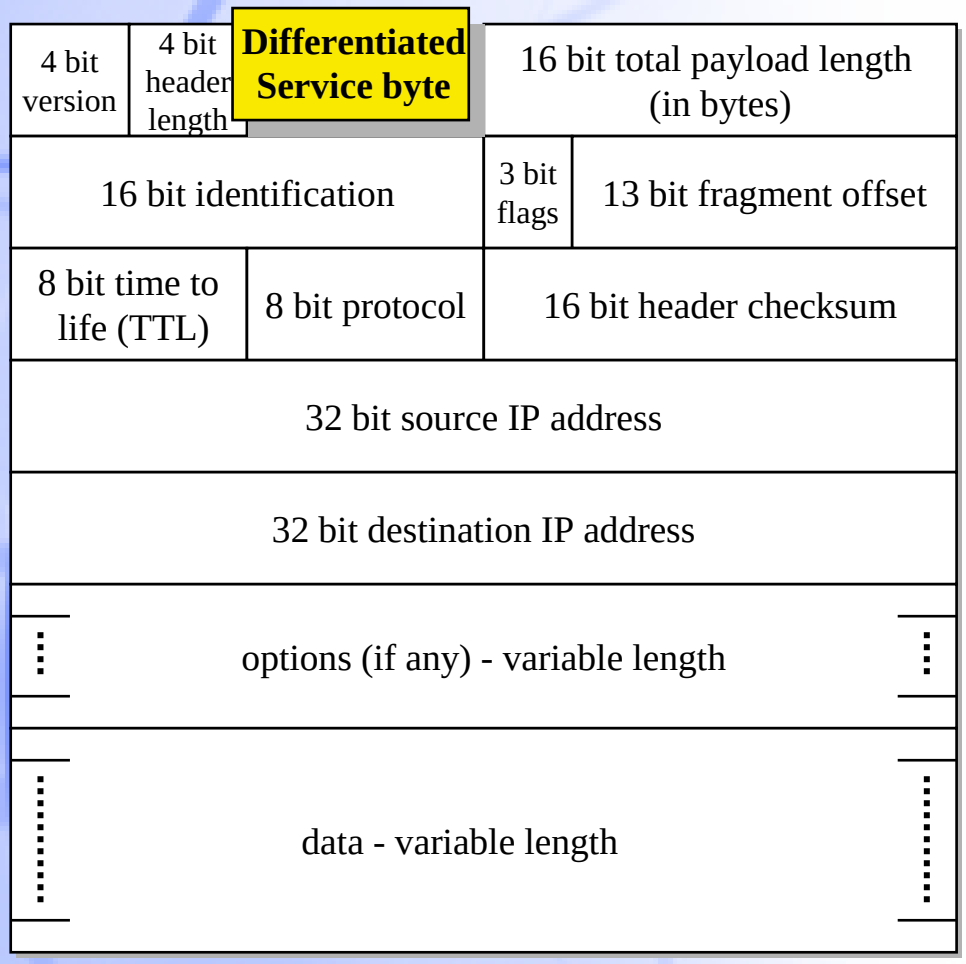
Traffic Conditioning, PHB (scheduling, buffer management, etc)

DS Node and Its Components



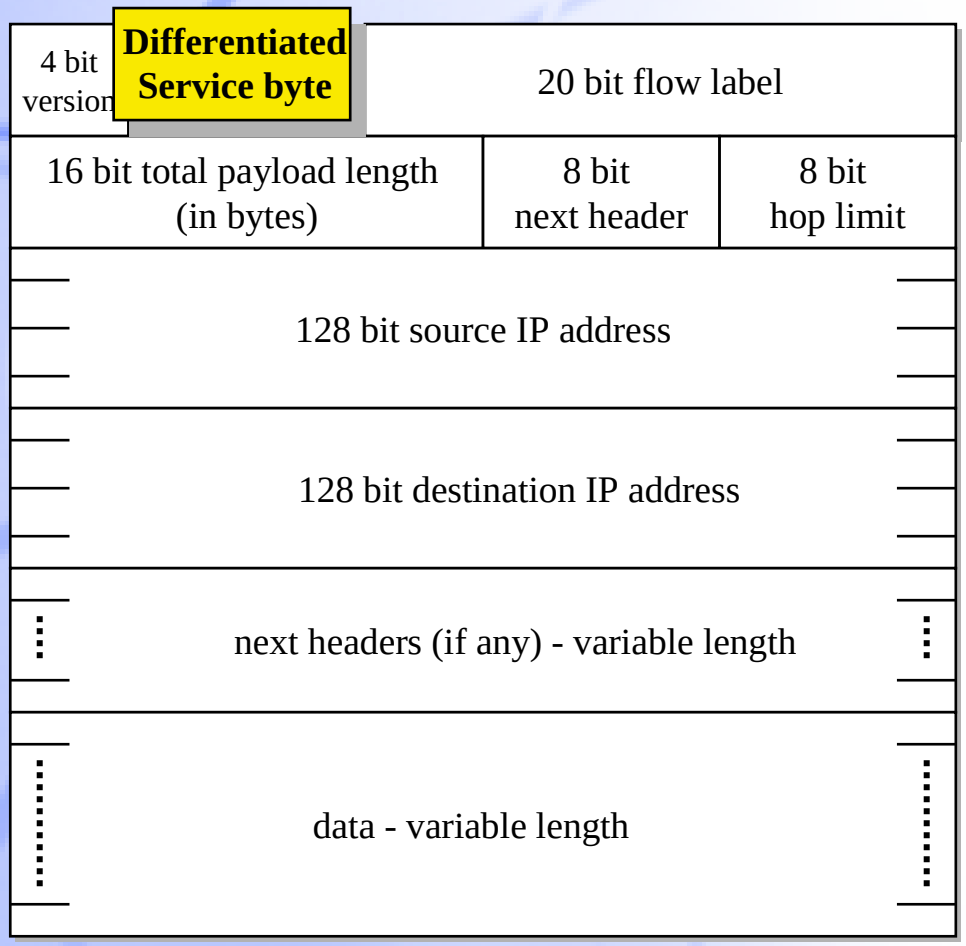
- The *Classifier* classifies the incoming packets (BA or BA/MF classification).
- The *Meter* measures the temporal properties against a traffic profile specified in the Traffic Conditioning Agreement (TCA).
- The *Marker* sets/remarks the DS field to a particular DSCP.
- The *Shaper* delays packets in order to shape the stream into compliance with the traffic profile.
- The *Dropper* discards packets in order to bring the stream into compliance with the traffic profile - known as “policing” of the stream.
- The *Traffic Profile* provides rules for determining whether the packet is within or out of this profile. It is part of the Service Level Agreement (SLA).
- These components are defined in RFC 2475.

Marking Field in IPv4



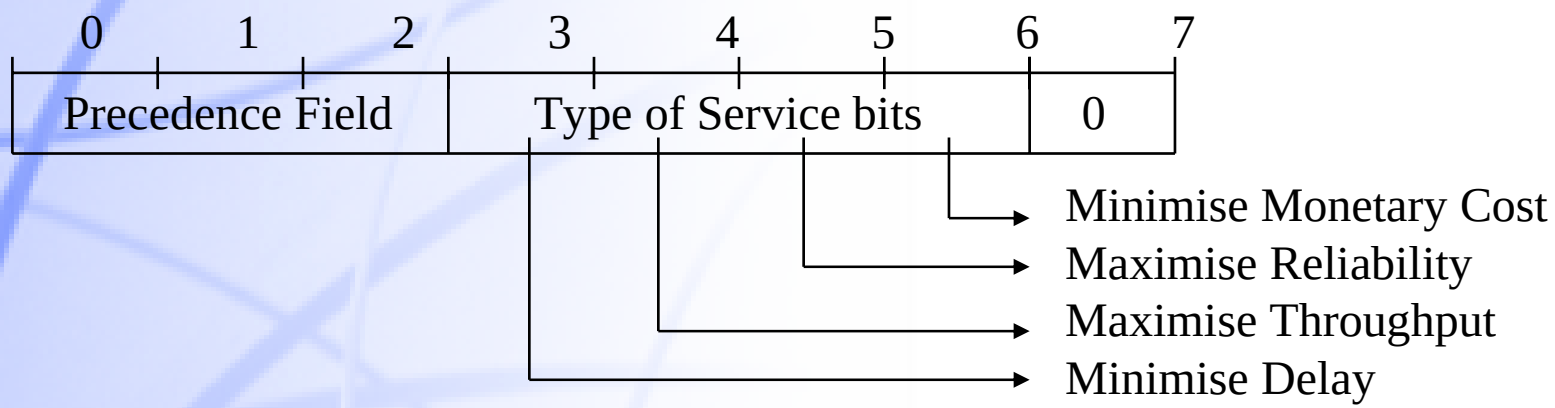
- **The former Type of Service (ToS) byte is replaced by the Differentiated Service (DS) byte.**
- **The byte will carry information on how the packet will be treated in a DS node.**

Marking Field in IPv6



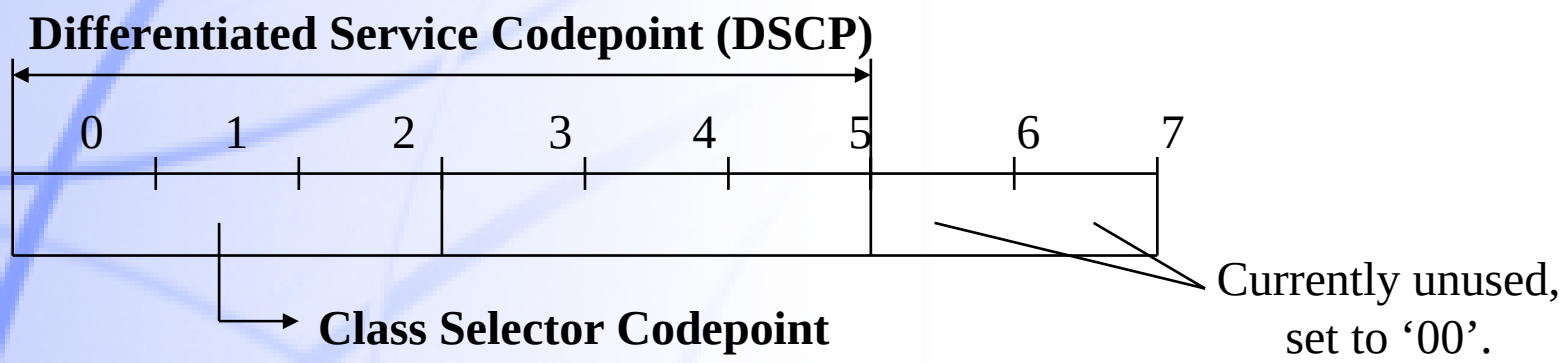
- The 4-bit priority field has been extended to 8 bit on the expenses of the flow label field (former 24 bits now 20 bit) and was then called Traffic Class byte.
- This Traffic Class byte is re-named to Differentiated Service (DS) byte.
- It has the same functionality as in IPv4.

Type of Service Byte



- **The Type of Service (ToS) byte offered the possibility to have QoS in the Internet but nobody used it, respectively implemented it in routers.**
- **Type of Service bits were used from some applications such as TELNET, FTP, SNMP, ... but not further processed in the network.**
- **Precedence field was used for network control traffic [Request for Comments (RFC) 1812].**

Differentiated Service Byte



- The Differentiated Services Codepoint (DSCP) defines the network service.
- The Class Selector Codepoint allows the mapping to the Precedence Field of the ToS byte.
- Codepoint space:
  - xxxxx0 ….. Standard Action
  - xxxx11 ….. Experimental or local use
  - xxxx01 ….. Experimental or local use, if necessary Standard Action allocation.
- The Differentiated Service byte is defined in RFC 2474.

Per Hop Behaviour (PHB)

- PHB is the externally observable forwarding behaviour to a DS flow in a DS capable node.
- Example: Packets with the DSCP = 101000, 101001, 101010 or 101011 will get the same treatment in a DS node.
  => The codepoints 1010xx (where x refers to 0 or 1) can be mapped to the same PHB.
- PHBs defined are:
  - Best Effort
  - Expedited Forwarding
  - Assured Forwarding

Best-Effort Service

- **The Best-Effort service is currently used in the Internet.**
- **DSCP is defined as the** *Default Codepoint* = '000000'
- **The service has the following properties:**
  - **No bandwidth guarantee.**
  - **No Quality of Service.**

Expedited Forwarding Service

- RFC 2598: "An Expedited Forwarding Service"
- Expedited Forwarding Service (EF) or Premium Service is understood as a Virtual Leased Line (VLL) service.
- It is defined by the Expedited Forwarding Per Hop Behaviour (EF PHB) with the recommended DSCP '101110'.
- The service has the following properties:
  - Peak bit rate - on flows or aggregated flows.
  - No bursts - only within the peak bit rate.
  - Low queuing delay - for real-time applications.

Assured Forwarding Service

- RFC 2597: "Assured Forwarding PHB Group"
- Assured Forwarding Service (AF) or Assured Service (AS) provides an exception of a certain amount of QoS to the flows and aggregated flows.
- The packets are labelled with a higher priority.
- The properties are:
  - Bandwidth exception but no guarantee.
  - Four AF Service classes with one queue each.
  - Three dropping precedence (Low, Medium and High).
  - Code point:

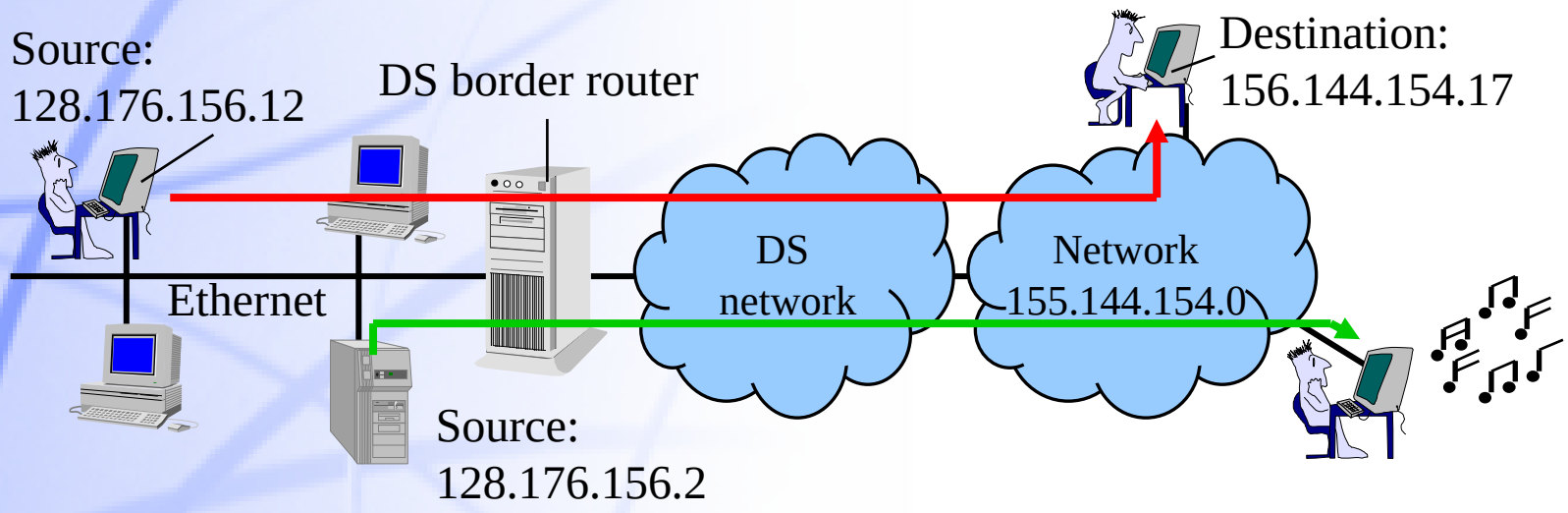
Drop Precedence	*Class 1*	*Class 2*	*Class 3*	*Class 4*
Low	001010	010010	011010	100010
Medium	001100	010100	011100	100100
High	001110	010110	011110	100110

Overview of the Class Selector Codepoints

- **All Class Selector Codepoints of the standardisation codepoint space (Bit pattern: xxxxx0) are used for the new services - yellow marked - and the old services.**

Service	*Class Selector Point*
Best-Effort service	000
AF Service Class 1	001
AF Service Class 2	010
AF Service Class 3	011
AF Service Class 4	100
EF Service	101
Network control traffic	11x (`x` refers to `0` or `1`)

Example of an IPv4-DS Profile Table



Source Addr.	Dest. Addr.	Source Mask	Dest. Mask	Proto-col	DSCP	Band-width (low AF)	EF-FIFO-depth/ medium AF-bandwidth
128.176.156.12	156.144154.17	255.255255.255	255.255255.255	0x06 (TCP)	101110 (EF)	150kb/s	15000 Bytes
128.176.156.2	156.144154.12	255.255255.255	255.255255.0	0x11 (UDP)	001010 (AF)	80kb/s	10 kb/s

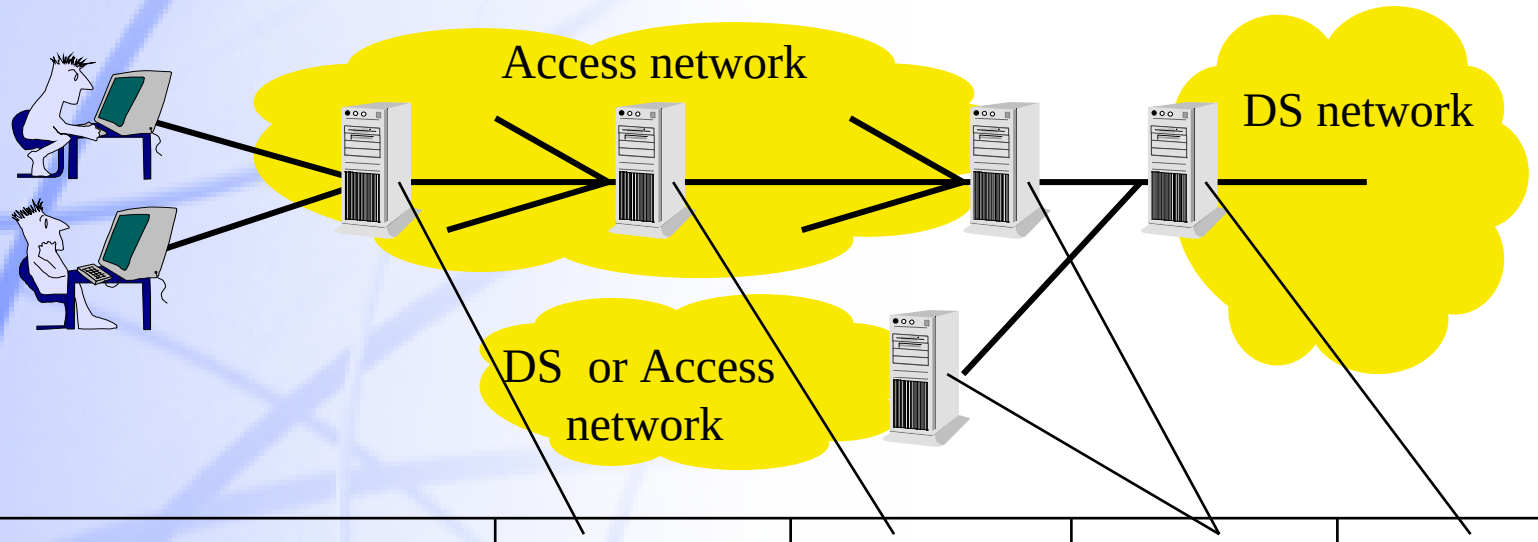
Classification Rules

- **Classification rules:**
  - **(*source packet address* AND source netmask)**
    EQUAL
    (table source address AND source netmask)
  - **(*destination packet address* AND destination netmask)** EQUAL
    (table destination address AND destination netmask)
  - ***packet protocol*** EQUAL **table protocol**
- **The amount of rules depends on the amount of involved fields.**
- **Depending if the rules are fulfilled, the packet will be marked or not.**

Network Service Parameters

- Each service has different parameters:
  - e.g. EF: Maximum bandwidth and depth of the FIFO, here 10 x MTU (Maximum Transmission Unit) of Ethernet (1500 Bytes, with the assumption that the Ethernet network has the smallest MTU through the network).
  - e.g. AF: Low and Medium Dropping Precedence bandwidth.
- Bandwidth parameters can be implemented by token buckets.
  - The packet will be processed depending on the state of these token buckets, e.g. the packet will be forwarded either discarded (EF) or re-marked (AF).

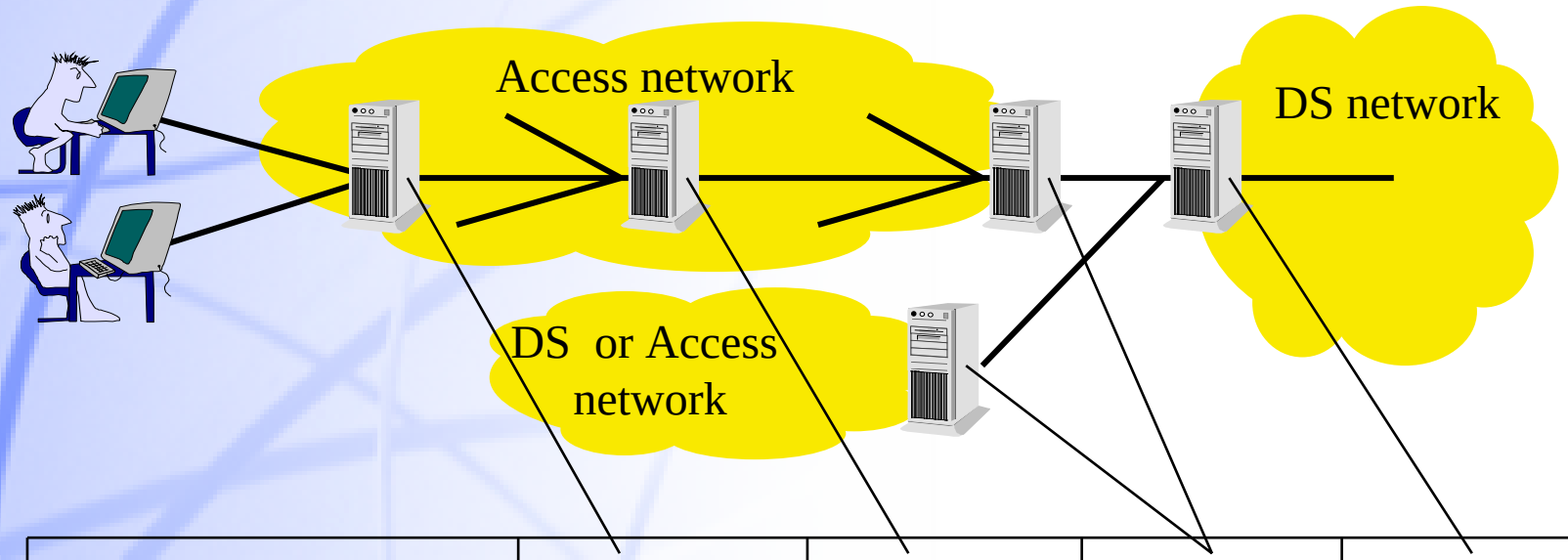
Components for the EF-PHB-Routers



	First Hop Router	*Interior Router*	*Egress Router*	*Ingress Router*
MF classification	X			X
BA classification	X (1)	X	X	X
Meter	X		X	X
Marker, Remarker	X		X	X
Policer/Dropper	X			X
Shaper	X		X	

(1) … in the case of trustable end-systems

Components for the AF-PHB-Routers



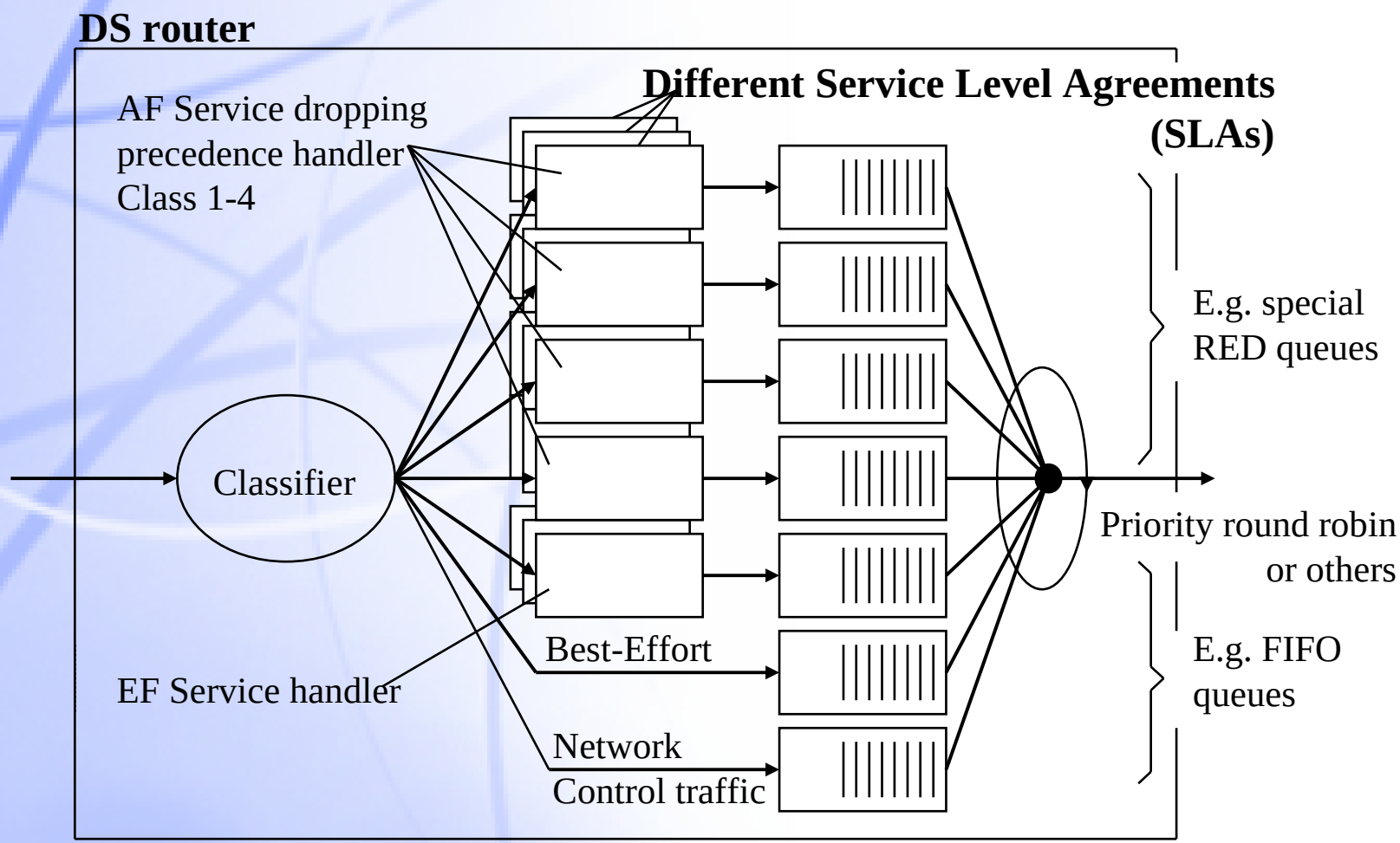
	First Hop Router	*Interior Router*	*Egress Router*	*Ingress Router*
MF classification	X			X
BA classification	X	X	X	X
Meter	X		X	X
Marker, Remarker	X		X	X
Policer/Dropper	X(2)	X(2)	X(2)	X(2)

(1) … in the case of trustable end-systems (2) ... in form of the AF Service queuing mechanism

DS Router Implementation (1)

- Each service has their own output queuing system.
- Expedited Forwarding Service and Assured Forwarding Service have connected service handler in front of the output queuing system.
- Four different kinds of traffic have to be classified:
  - Expedited Forwarding Service traffic.
  - Assured Forwarding Service traffic with sub-classification.
  - Best-Effort Service traffic.
  - Network control traffic.
- After classification of the traffic, the Best-Effort traffic and the network control traffic will be forwarded without any manipulation in their output queuing system (e.g.: First-In-First-Out queuing or others).

DS Router Implementation (2)

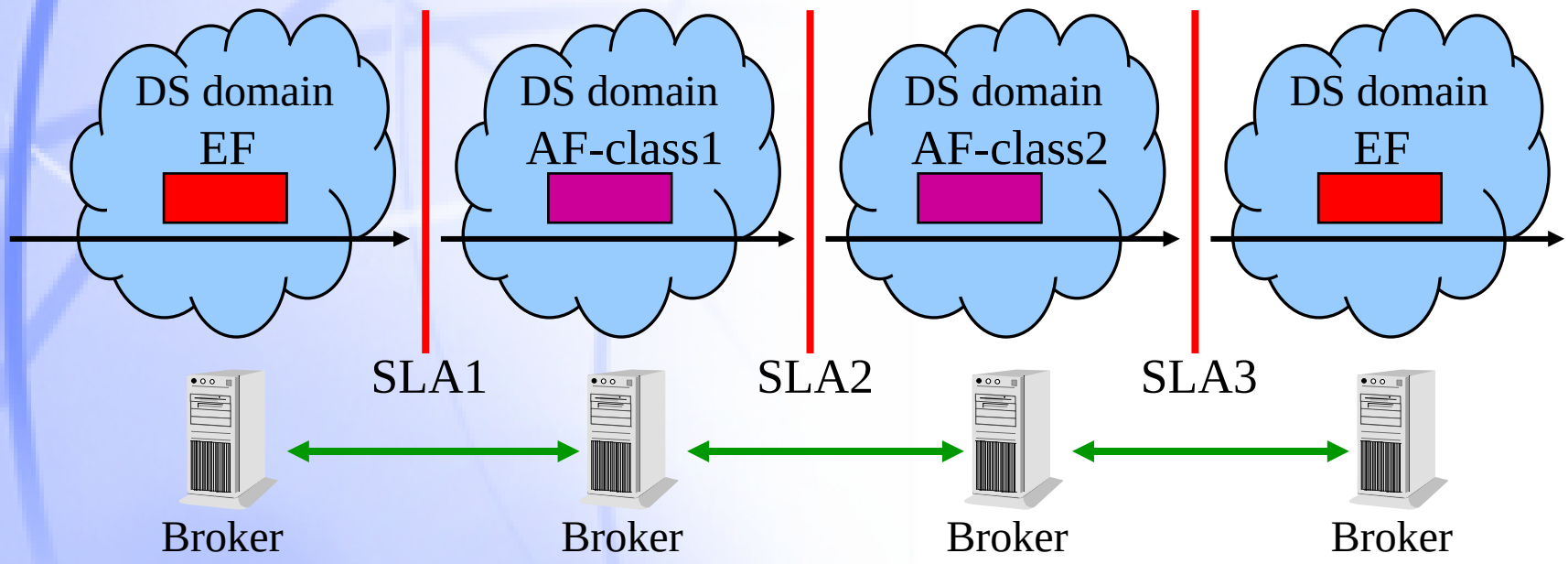


Service Level Agreement

- It is a contract between two or more Internet Service Providers (ISPs).
- All parties agree that the traffic should be within a certain profile.
- Traffic out of the profile will be either dropped or re-marked.
- This profile can be used for traffic charging.
- For access networks, the SLA is on flow level.
- For backbones, the SLA is on aggregated flow level and also on flow level for particular flows.
- The SLA contains policing parameters such as
  - token bucket size,
  - burst size,
  - queuing delay
  - ...

Bandwidth Broker and Heterogeneous Networks

- Bandwidth brokers offer the bandwidth brokerage services - the negotiation of SLAs.
- Since we will not have a homogeneous network structure, bandwidth brokers are necessary.



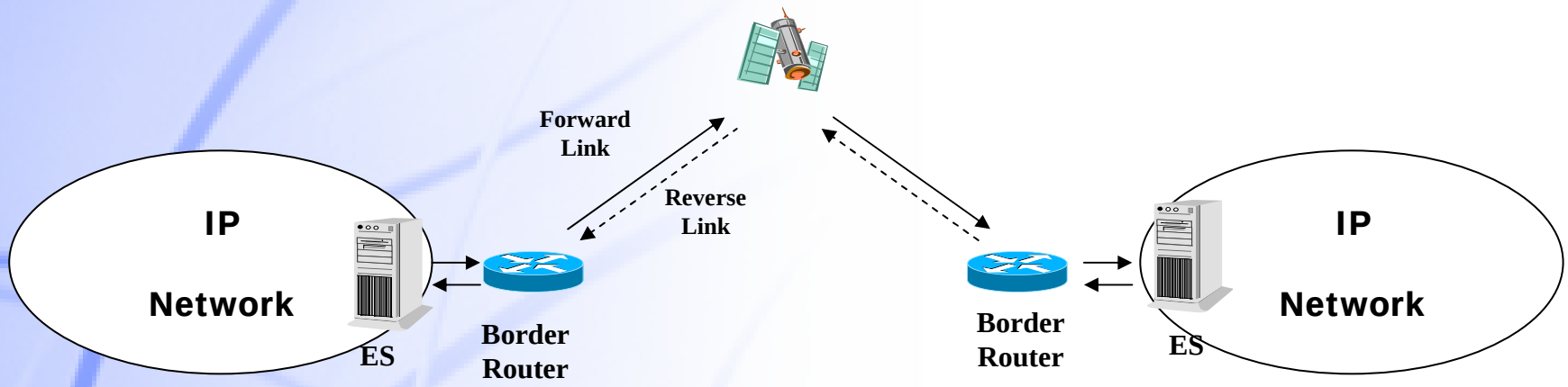
Comparison of Intserv & Diffserv Architectures

	Intserv	Diffserv
Granularity of service differentiation	Individual Flow	Aggregate of flows
State in routers(e.g. scheduling, buffer management)	Per Flow	Per Aggregate
Traffic Classification Basis	Several header fields	DS Field
Type of service differentiation	Deterministic or statistical guarantees	Absolute or relative assurance
Admission Control	Required	Required for absolute differentiation
Signaling Protocol	Required(RSVP)	Not required for relative schemes

Comparison of Intserv & Diffserv Architectures

	Intserv	Diffserv
Coordination for service differentiation	End-to-End	Local (Per-Hop)
Scope of Service Differentiation	A Unicast or Multicast path	Anywhere in a Network or in specific paths
Scalabilty	Limited by the number of flows	Limited by the number of classes of service
Network Accounting	Based on flow characteristics and QoS requirement	Based on class usage
Network Management	Similar to Circuit Switching networks	Similar to existing IP networks
Interdomain deployment	Multilateral Agreements	Bilateral Agreements

Satellite Link



- Long Propagation Delay
  - LEO: 20-25 mseg
  - MEO: 110-150 mseg
  - GEO: 250-280 mseg
- High Bit Error Rates (BER)
- Asymmetrical Link

TCP implications on Satellite

- Slow Start procedure:
  - Long RTT for ACK
  - Reduce Rates for increasing window
  - Increase duration for slow start
  - Hurt interactive and MM applications
- Asymmetric channel:
  - Forward Link (FL): High Bandwidth
  - Reverse Link (RL): Low Bandwidth
    - Delay of ACK, worse in case of errors
- BER
  - Errors managed by FEC (Turbo Codes)
  - In TCP errors are manage as congestion reducing the windows

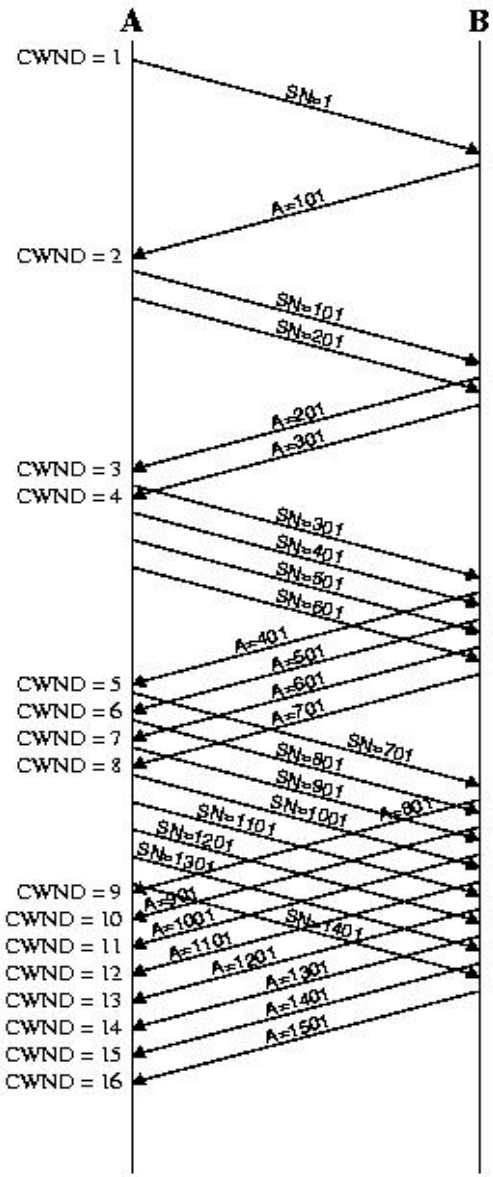
TCP/IP Protocol

Management Procedures:

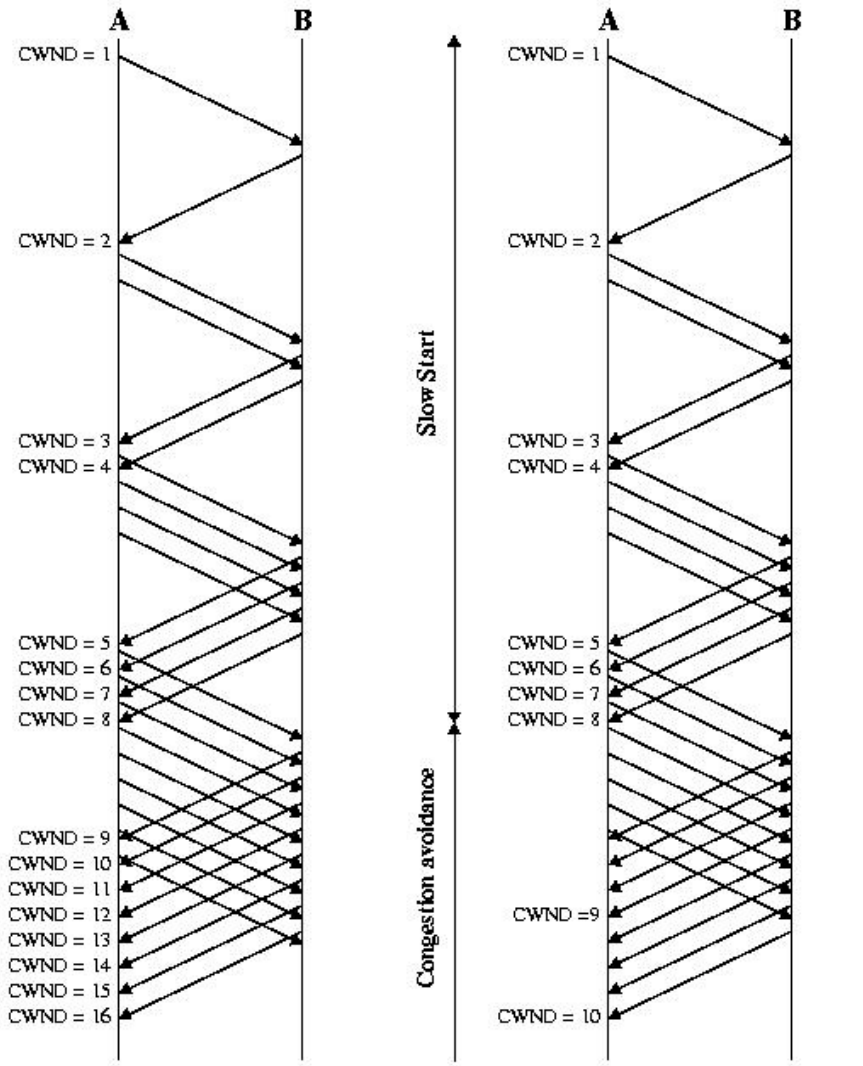
- **Slow Start**
  - TCP sender create a congestion windows (cwnd): max. segments sender is going to send
  - cwnd=MIN[cwnd, credit]; credit=Receiver Buffer
  - Initially cwnd=1
  - Increase by each ACK received

- **Congestion Avoidance**
  - Sender send more that Network can deliver.
  - assume all losses are caused by congestion
  - implemented together with slow start
  - In case of congestion (duplicate ACK or timeout)
    - ssthresh= cwnd/2
    - cwnd=1 and slow start until cwnd=ssthresh
    - If cwnd=ssthresh then cwnd increase by 1 each RTT

Slow Start


A
B
CWND = 1
SN=1
A=101
CWND = 2
SN=101
SN=201
A=201
A=301
CWND = 3
CWND = 4
SN=301
SN=401
SN=501
SN=601
A=401
A=501
A=601
A=701
CWND = 5
CWND = 6
CWND = 7
CWND = 8
SN=701
SN=801
SN=901
SN=1001
A=801
SN=1101
SN=1201
SN=1301
CWND = 9
CWND = 10
CWND = 11
CWND = 12
CWND = 13
CWND = 14
CWND = 15
CWND = 16
A=901
A=1001
A=1101
A=1201
A=1301
A=1401
A=1501
SN=1401

Congestion Avoidance



(a) Slow start, ending with a timeout (b) Slow start followed by congestion avoidance

ssthresh=8

TCP/IP Protocol

- Fast Retransmit
  - When ACK is not received: Lost or Reorder ?
  - Wait until 3 retransmission ACKs to perform a retransmission
- Fast Recovery
  - slow start will be taken over by congestion avoidance after fast retransmit

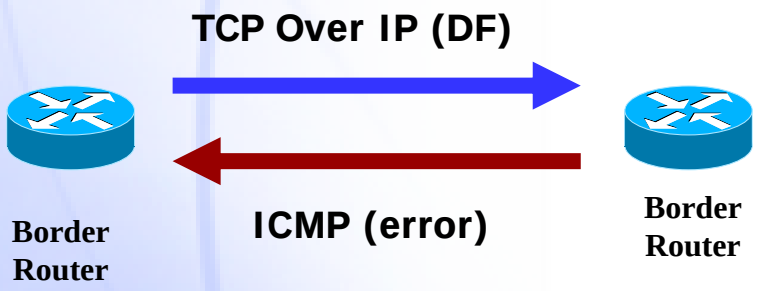
Enhancement on TCP/IP

- TCP/IP design for cable links, problems with wireless connections.

- Path MTU Discovery
- Larger congestion window (*cwnd*)
- Larger Slow Start Threshold (*ssthresh*)
- Selective Acknowledgements

Path Maximum Transfer Unit (MTU) Discovery

- to discover the maximum segment size without being subjected to IP fragmentation
- Use larger segments means send more effective data over control data.
- Larger (optimal) segments increase cwnd
- Expected delay in case of probing all the gateways with different segment size.



Larger Congestion Window (*cwnd*)

- the increment of congestion window (*cwnd*) depends entirely on the receipt of ACKs
- We can increase the default initial size of *cwnd*, there will be more segments in the startup
- this will trigger more ACKs and hence speed up the growth of *cwnd*
- the exact size of the *cwnd* is still an open issue but there is a proposal of **4 segments**[1] during the startup

[1] RFC 2488: "TCP/IP enhancements for Satellite Networks". IEEE Communications Magazine. July 1999

Larger Slow Start Threshold (*ssthresh*)

- the default size of *ssthresh* is 65535 bytes (64 kbytes)
- with the increment of *ssthresh*, the issue of preventing the growth of *cwnd* no longer exists
- larger *ssthresh* will make sure the buffer in receiver's window can accommodate high throughput of the link
- Buffer in the received be design properly to avoid overflow.
- Some proposal:

$$ssthresh = ((B + UT)/2)$$

B : Buffer Size

U : Bandwidth

T : Round Trip Time (RTT)

Selective Acknowledgements (SACKS)

- Allow TCP receivers to inform TCP senders exactly which segment has arrived
- this will enable TCP to recover from lost segments faster avoiding needless retransmissions
- the sender is able to determine which segment need to be retransmit following a loss detection
- this also allows the sender to retransmit and at the same time transmit new segments

Quality of Service (QoS)

- Satellite QoS by means of:
  - ATM, but problems to be supported
  - Normally TCP/IP over PPP links in satellite
- We will concentrate on IP-based QoS Techniques:
  - Best Effort Service
  - Resource Reservation Protocol (RSVP)
  - Differentiated Services (DiffServ)
  - Multi Protocol Label Switching (MPLS)
  - ….

Differentiated services (DiffServ)

- **IP header**: DCSP set accordingly type of applications/traffic
  - Real Time Application
  - Non Real Time Application
  - ....
- **Service Level Agreement(SLA)** by ISP:
  - Premium Service
  - Assured Service
  - Olympic Service

Effects of DiffServ on Mitigated TCP/IP

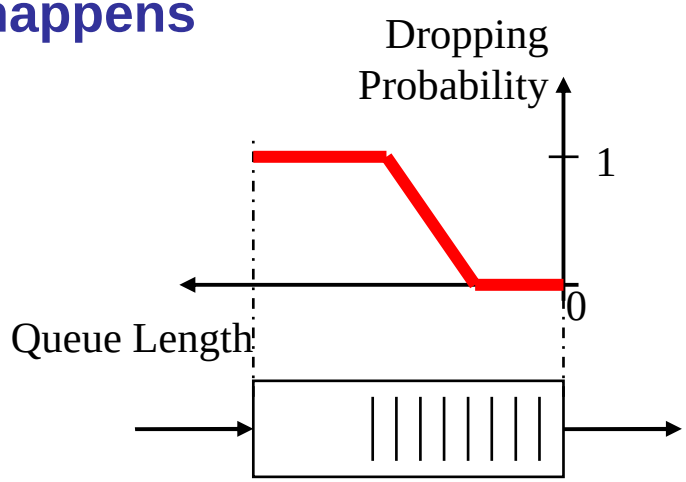
- Potential mitigated TCP/IP offers direct impact on the throughput of satellite channel but regardless of the classes of traffic
- therefore, DiffServ offers an improvement over how to classify and shape different classes of traffic according to their priority

Effects of DiffServ

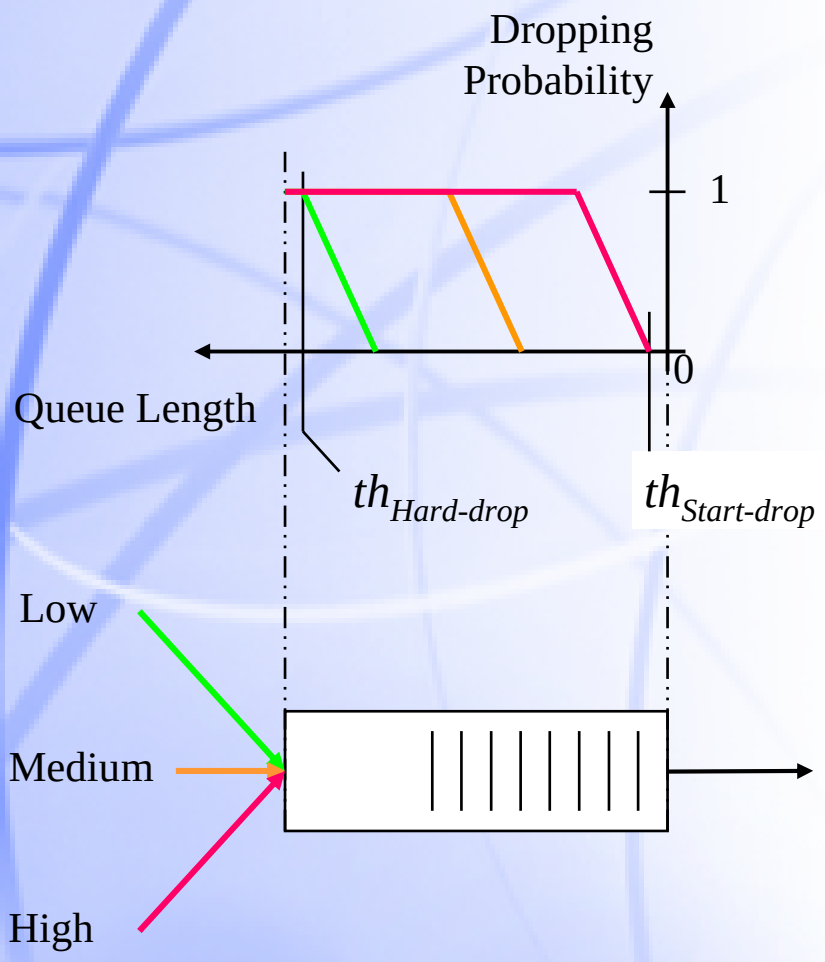
- Two main services:
  - **Premium Service:**
    - **can potentially overwhelm the satellite link by using all the available bandwidths**
    - **can be overcome by limiting the percentage of bandwidth allocated for Premium Service**
    - **no guarantee QoS due to high delay and high BER**
  - **Assured Service:**
    - **utilizes Random Early Detection(RED) queue management scheme**
    - **RED can be incorporated with TCP/IP protocol**
    - **RED for In and Out(RIO) is a more advanced RED**

Random Early Detection

- The Random Early Detection (RED) algorithm computes the probability to discard a packet which just arrives at a queue.
  (for more details, please see: S. Floyd and V. Jacobson, "Random Early Detection Gateways for Congestion Avoidance," in IEEE/ACM Transactions on Networking Vol.1 N.4, pp. 397 - 413, August 1993.)
- prevent queue overflow within the router
- drop packets randomly before overflow happens
- RIO (RED for In and Out)
  - a more advanced RED
  - maintain two RED algorithms for
    - In packets (threshold)
    - Out packets (threshold)
  - when size of queue is in between the two thresholds, only the out packets are drop



Assured Forwarding Service Queuing



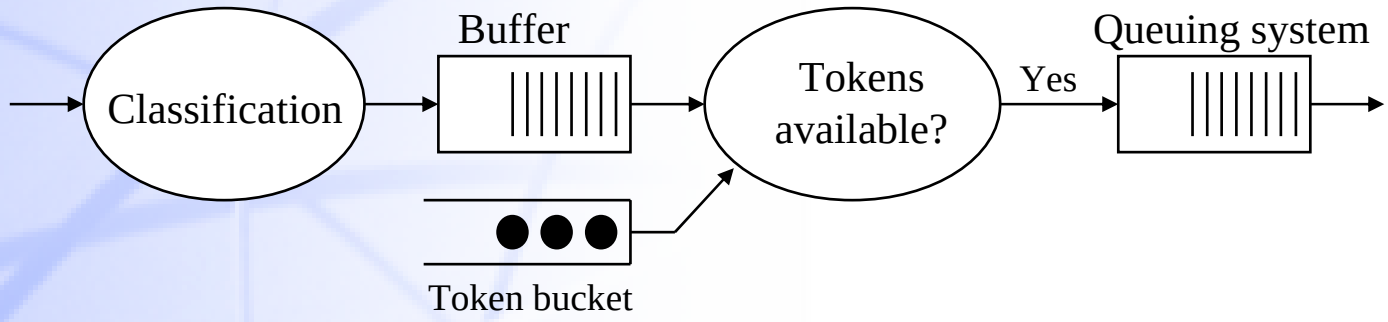
- **Dropping probability of each precedence will be computed by an independent Random Early Detection (RED) algorithm.**
- **For each class, one queue is implemented to avoid re-ordering of the flows which belong to the same AF service class.**

Expedited Forwarding Service

- Depending on the node location, Expedited Forwarding Service flows can be controlled by two mechanisms:
  - Shaper
  - Dropper
- The bandwidth control mechanism can be implemented by a token bucket.
  - The token bucket size should be equal or bigger than the largest possible IP packet.
  - Which layers will be involved in the size measurement depends on the SLA (e.g.: only the IP packet, or IP packet and the header of the link layer, etc.).
- Only one output queuing system is required. This can be a normal FIFO queue or a RED queue or another queuing system.

Expedited Forwarding Service Shaper Mechanism

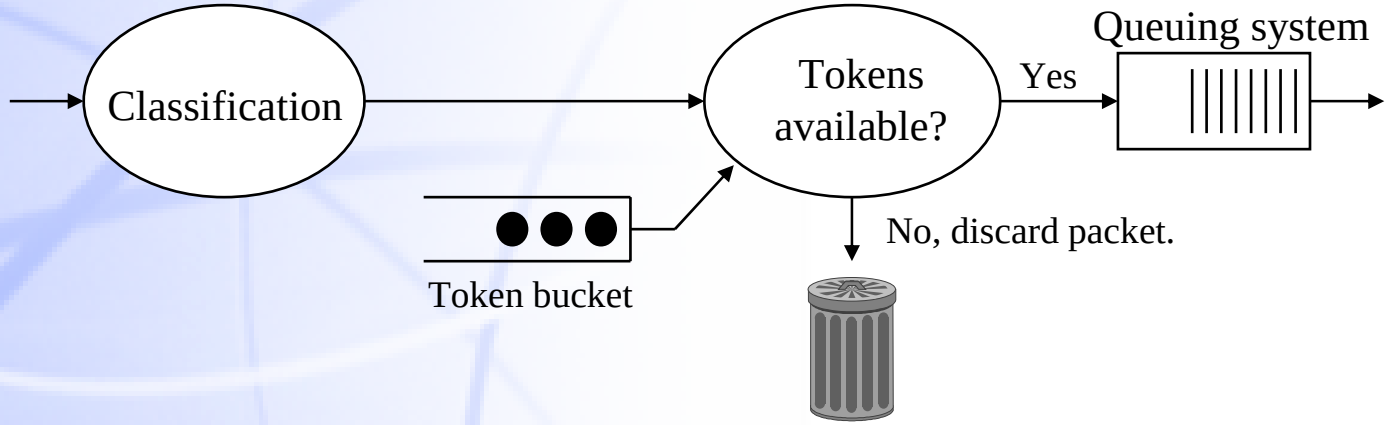
EF Service handling in Egress and First Hop routers



- **An EF Service packet will be stored until the First-In-First-Out buffer overflows.**
- **If tokens are available, the buffer will be emptied in relation to the amount of tokens.**
- **The token bucket will be updated after reading the buffer.**

Expedited Forwarding Service Discarding Mechanism

EF Service handling in Ingress routers



- **An EF Service packet will be forwarded immediately, if tokens are available otherwise it will be discarded.**
- **The token bucket will be updated after forwarding the packet.**

Impact of DiffServ on Path MTU

- with the Path MTU segment being given a higher priority, the process of determining the highest segment size will be quick
- from an experimental result, the optimal value for 128 kbytes window size is 1024 bytes.
- recommended to dimension the minimum MTU to 1024 bytes

Impact of DiffServ on Slow Start and Congestion Avoidance

- RED/RIO will act intelligently to discard excessive segments in the router
- the chance of premature trigering of slow start and congestion avoidance due to segment loss will be less for higher priority application (ej. Real Time Applications)

Impact of DiffServ on Congestion Window Growth

- due to DiffServ prioritization, real-time application will expect a faster growth in *cwnd*
- this will enable a better utilization of bandwidth in terms of reducing wastage and improving quality for different kind of applications
- when the window size of 16 kbytes is increased to 64 kbytes, there is an improvement of 35%
- from 64 kbytes to 128kbytes, there is only a slight improvement (limitation due to hardware characteristic)
- the maximum *cwnd* of 128 kbytes can be used as a benchmark for future performance studies

Impact of DiffServ on Selective Acknowledgements

- It only offers an indirect mitigation towards the returning path
- ACK segments with higher priorities will be transmitted ahead of other traffic
- SACK will further mitigate the throughput by ensuring only selected ACKs are returned, hence reducing the number of ACKs in the reverse path

Conclusions

- DiffServ + QoS Brokers (BB) is a good solution for providing MM requirements.
- Important to clasify traffic based on agregation, even BE traffic
- TCP/UDP protocol performance over satellite must be consider including improvements.
- Relevant heterogeneous scenarios terrestrial-satellite with IP as basic
- Network selection problem addressing today by IETF.

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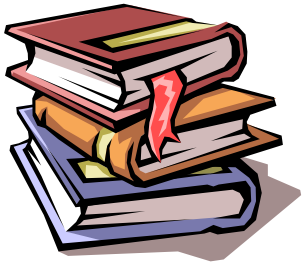

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