IP Quality of Service

Jukka Manner, TKK

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What is IP Quality of Service?

In this presentation, IP QoS means:

The technology, information and decisions used in routing IP packets in order to fulfill the expectations of clients on the quality of the IP connectivity.

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Why do we need to talk about IP Quality of Service?

If the network had unlimited capacity, there would be no need to talk about "Quality of Service": everybody could send and receive as much as they want.

However in practice there is no such thing as unlimited capacity.

Contents

- Part I: The why and how of providing IP QoS, and what applications can do themselves
- Part II: Mechanisms for providing service to packets on a “case-by-case” basis (queuing mechanisms, filters, schedulers, …)
- Part III: (IETF) architectures that make use of these mechanisms to provide QoS to end-user applications

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Part I: The why and how of IP QoS

Routing in the Internet

- The default service of the Internet is "best-effort"; no particular special effort is put in forwarding packets
- When a packet is received its destination is compared to the routing table and the packet is put to the queue of the output interface; only the destination address matters
- Each packet is served in the same way in each router
- This is an equal service between competing flows
- As the load varies, the network's service varies
- Any packet may be dropped when congestion occurs
- During and after congestion, senders are expected to adjust their packet sending rate

New Applications and Usage

- The load in the network is very heterogeneous
  - The main load is HTTP/FTP/email traffic based on TCP
  - Streaming multimedia applications are gaining popularity, e.g. network radio, streaming video, VoIP, etc.
  - Also, emergency calls, network control signalling, database queries for E-commerce, etc. are increasing
- Many of these need reliable and stable forwarding services, which means a need for steady transmission delay and low packet loss
- Requirements can be about bandwidths (10 Kbps up to Mbps), low packet loss, low delay, etc.
- The fundamental problem is that no dedicated circuit-switched connections exist in a packet switched network
- Thus, no dedicated resources can be allocated to single flows

How to support IP QoS

- Most simple form of IP QoS is over-provisioning
- Applications can try to adapt the data transfer to the network capacity, and congestion situation, e.g. TCP, DCCP, RTP (Note the difference between application and transport level)
- Applications can duplicate the payload, e.g.,use FEC, and hope that the receiver can rebuild the original data
- Admission control schemes can be used to prohibit new flows from entering a network:
  - Distributed algorithms base the decision on the knowledge of the load in the network
  - Explicit signaling protocols can carry requests of hosts to let the network know, e.g., the bandwidth needed by a new flow
- IP routers can forward incoming packets differently based on the sender/receiver, the type of flow, an explicit request, etc.

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The Real-time Transport Protocol (RTP)

- With lack of QoS mechanisms in the Internet, how can audio-video applications deliver an adequate service?
- TCP is an earlier example of an adaptation mechanism:
  - When packets get lost, slow down the sending rate, since there ”must” be congestion in the network
- Ordinary multimedia flows use commonly a fixed codec and send a constant stream
- When packets get lost, the receiver notices a jerky and mismatched audio/video; delayed packet are as well “lost”
- No way for the sender to know about network congestion and change the transmission rate
- Thus, try to adapt yourself, e.g., lower the quality of the stream

The Real-time Transport Protocol (RTP)

- RTP is based on three components:
  - A family of codecs to code the multimedia flow, ranging from GSM-codecs to MP3, MPEG4 (over 40 payload formats, so far)
  - A transport protocol operating over UDP to deliver the audio/video information
  - A control protocol (RTCP) to control the media delivery and provide feedback to the sender: received/lost packets, timestamps, jitter, etc.
- RTP specifies payload formats, security, header compression, multiplexing of streams, etc.
- Codecs have different delay and bandwidth requirements
- The feedback provided to the sender allows it to change the stream properties and, thus, change its requirements from the connecting network

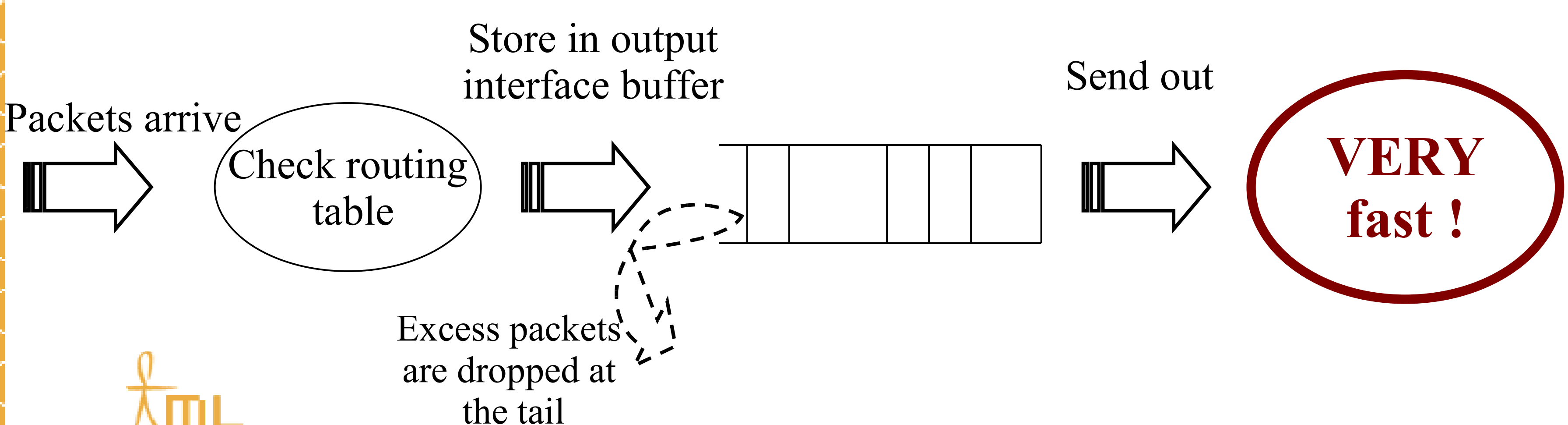
The Datagram Congestion Control Protocol

- Many applications need to send a constant stream of packets
- So far the only option has been to use UDP, which does not have flow and congestion control algorithms, or RTP/RTCP
- If the network becomes congested, TCP-based flows will react, slow down their transmission rate – UDP flows won't
- If applications want to support congestion control, they must implement their own messaging and algorithms
- The Datagram Congestion Control Protocol (DCCP) is an effort to define a "congestion-aware UDP"
    - Still an unreliable datagram transport protocol
    - Receiver sends feedback to the sender on how packets are received
    - Sender adjust the sending rate based on the feedback
    - Different congestion control algorithms can be used, currently two are defined: TCP-like and TCP-friendly algorithms
    - Still, does not really work welll with actual streaming applications

Part II: Mechanisms for IP QoS

The Simple Mechanisms for IP Routing

- Three components affect the packet handling in a router:
  - The routing table lookup,
  - The packet queuing mechanism, and
  - The packet scheduler at the output interface
- The most simple combination:
  - First-in-First-out scheduling and tail drop
  - Problem: cannot provide service distinction



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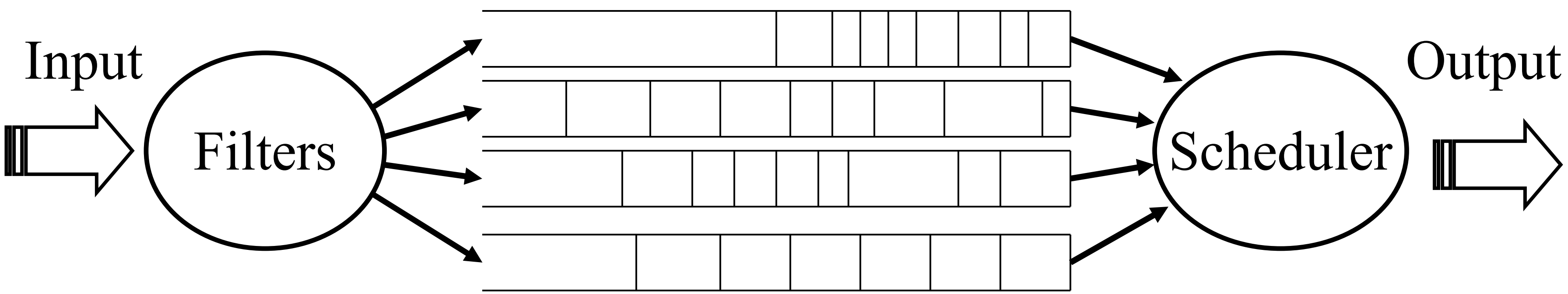
Distinguishing Flows

- A different service level can only be accomplished if
  - We can identify certain packet flows, and
  - Forward these separated from other flows
- In practice we need filters than can distinguish certain flows and more than one queue
- Filtering is most commonly based on protocol header fields
- Different scheduler principles can be used for the queues



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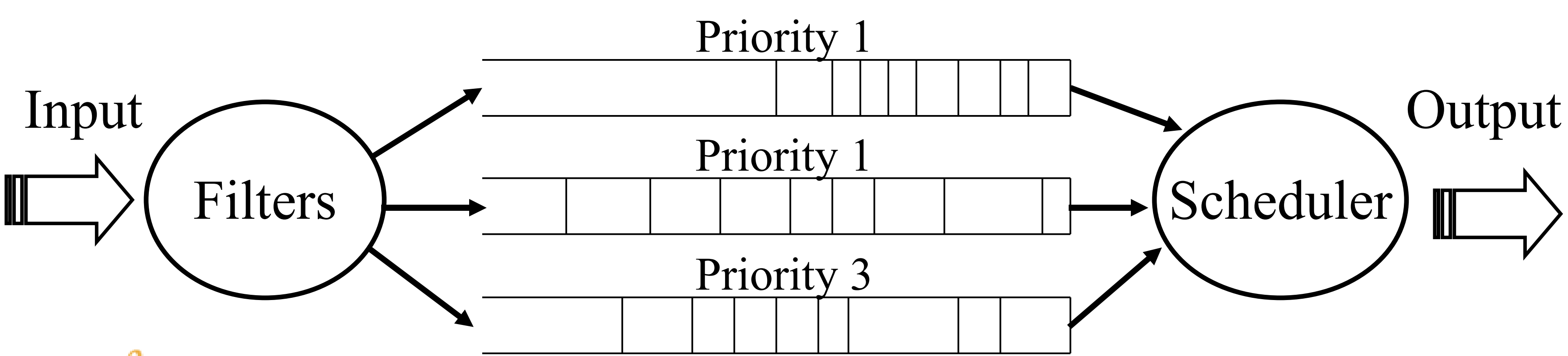
Most Common Schedulers (PRIO)

- A straightforward scheduler uses priority queuing:
  - Packets are filtered and queued in several queues
  - Packets in highest priority queue (1) are all sent before any packets from the next highest queue (2), etc.
- Issues:
  - Very good service to the highest class of packets
  - Very unfair to lower priority queues
  - Easily leads to starvation in the lower priority queues



Most Common Schedulers (CBQ)

- A variation of priority queuing is Class Based Queuing:
  - Also called Weighted Round Robin
  - Queues are assigned relative priorities
  - During a time interval t each queue is emptied an amount relative to its priority, for example:
    - Queue 1: 8 pcks, 8/15 (53%) of link BW
    - Queue 2: 4 pcks, 4/15 (27%) of link BW
    - Queue 3: 2 pcks, 2/15 (13%) of link BW
    - Queue 4: 1 pck, 1/15 (7%) of link BW

But what if…
(60B audio x 8 = 480B)
(120B HQ audio x 4 = 480B)
(540B HQ video x 2 = 1080B)
(1500B FTP (TCP) = 1500B)

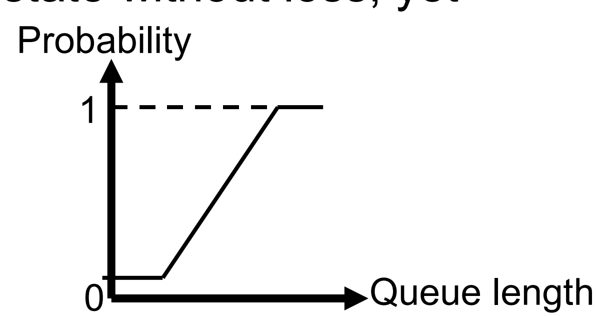
- Issues:
  - More fairness between different queues, avoids total starvation
  - Major concern is that the amount of data taken from a queue is based on packets
  - Also, setting up the queues requires many parameters (Linux: over 20)

Most Common Schedulers (WFQ)

- More complex scheduler: Weighted Fair Queuing
- Similar to CBQ but uses more information and calculations to decide which packets to forward next
- Estimates each packets time to be sent during a round and sends packets in that order
- Issues:
  - Accurate link sharing based on weights
  - Provides high level of fairness to different classes
  - Bounds delay (Parekh-Gallager) but not jitter
  - Consumes much more cpu cycles to calculate the next packet to handle
  - May lower the maximum throughput of the link
  - Should not be used if there are numerous queues

Queuing Strategies

- Most typical is tail-drop: when congestion occurs, buffers are full and arriving packets are (all) dropped
- Should take precautions to not run out of buffer space
- Various active queue management mechanisms can be used to drop specific packets, not always the last arrived
- Random Early Detection (RED) is one solution:
  - Packet drop is based on queue length: larger queue, higher probability of an incoming packet to get dropped
  - RED is fair: probability of a flow’s packet to be lost is proportional to its share of link bandwidth
  - Can also mark packets instead of dropping them (IP ECN bits), allowing sources to detect network state without loss, yet
  - Numerous variations of RED



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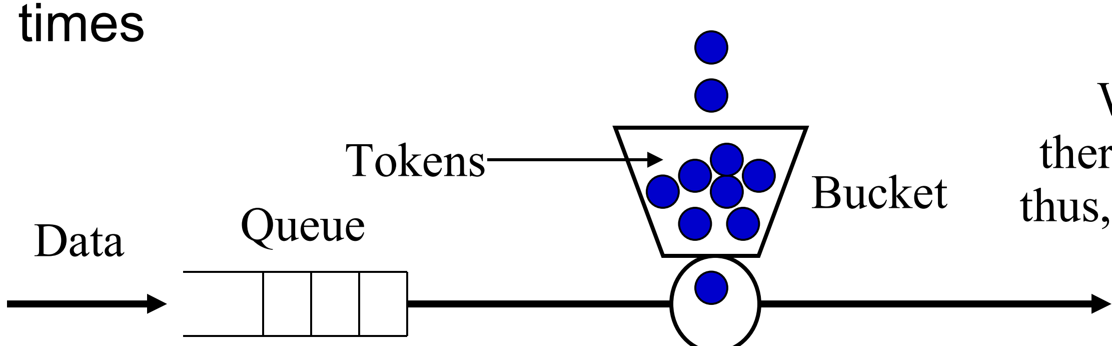
Additional Mechanisms

- Leaky bucket and token bucket filters
- Data packets ”leak” from a bucket of depth ***n*** with a rate ***R***
- Token bucket filter (TBF) adds “tokens” and a “bucket”:
  - Tokens are provided at a fixed rate ***R***
  - The bucket can accommodate some tokens for future use allowing bursts of packets
  - Still, the filter only allows a maximum rate ***R*** to flow through
- A number of derivate of WFQ and CBQ exist (Linux: HTB)
- Also pricing can be used to affect load in classes and at peak times



With leaky bucket,
there is no actual bucket,
thus, no bursts are possible

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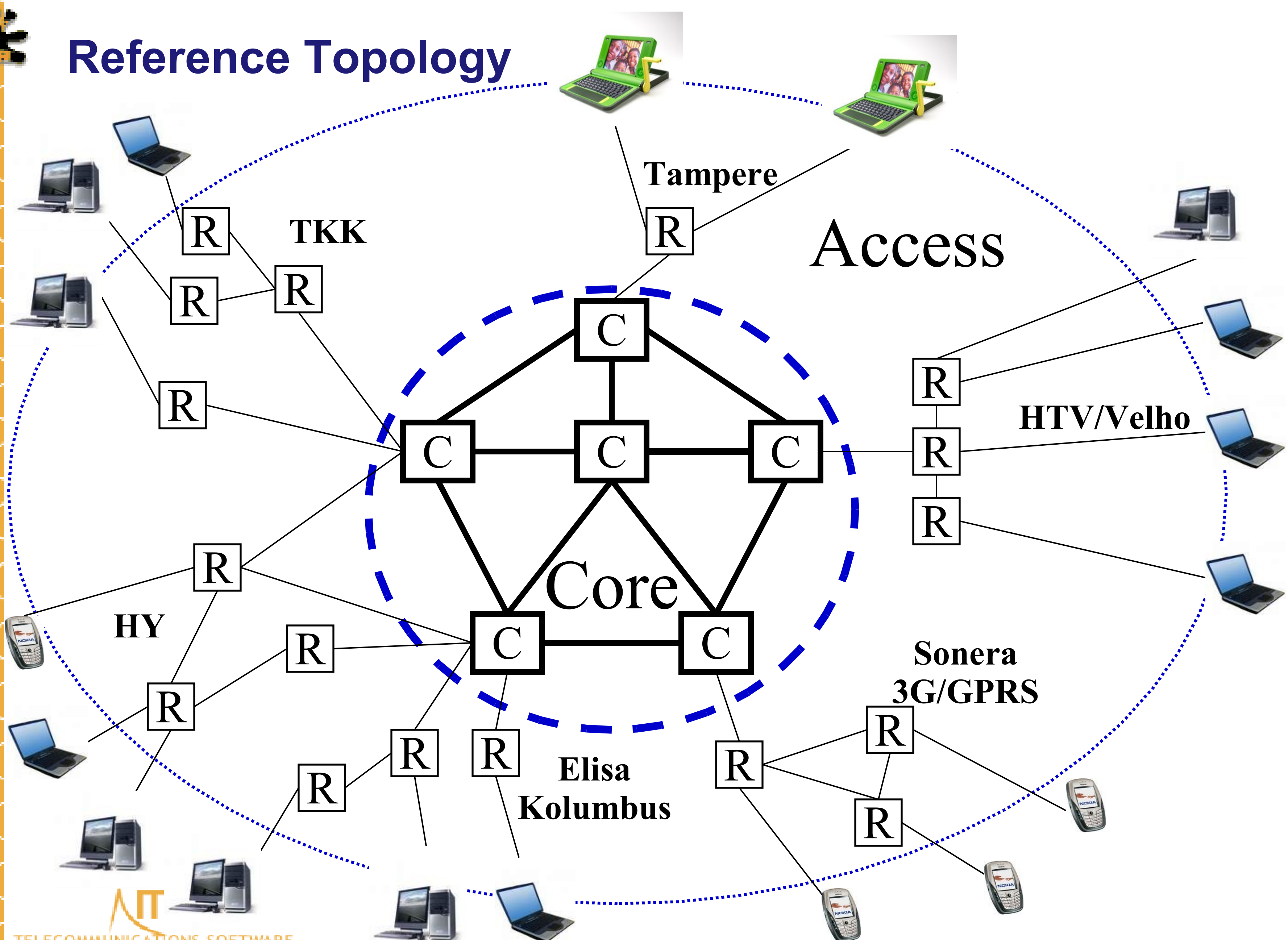
QoS Routing

- The default router forwarding does not take into account resource availability
- With a broader collection of links, we would need some way of finding various paths that would support the traffic
- Also in view of network efficiency, alternate QoS-enabled paths would be beneficial
- The routing protocol needs to include information about "resources", and
- The routing decision in a router needs to exploit this information to find the "best" next hop for each packet
- Still, must not often shift directions of individual flows: if the characteristics of the path change, transport protocols react

Where to Implement QoS?

- Link speeds faster towards the core; more traffic
- Access routers generally do not have to handle high packet switching rates and, thus, can do complex traffic filtering, classification and policing
- The overhead of implementing QoS in the core would affect a large amount of traffic and lower the throughput
- Thus, access networks could do per-flow packet handling, and
- Backbones forward packets in packet aggregates, a limited number of classes (or even FIFO)
- Note that the quality of the end-to-end connection is as good as the weakest link

Reference Topology



Part III: QoS Architectures

Objectives of QoS Architectures

- To control the network service response such that:
  - The response to a specific service is consistent and predictable
  - A client is provided with a level equal to or above a guaranteed minimum
- To allow a client to request in advance a service response
- To control the contention for network resources such that:
  - A client can be provided with a superior level of service
  - A client does not obtain unfair allocation of resources
- To allow for efficient total utilisation of network resources, while providing a range of different network services
- Quality can be bandwidth (min, max, average), delay, jitter, packet loss rate, etc.

IETF QoS Architectures

- Numerous proprietary architectures exist within the academic community and the telecommunications industry, e.g. YESSIR, INSIGNIA, RMD, Mobile RSVP, Localized RSVP
- IETF has two main, although very different, standards:
  - The Integrated Services architecture (IntServ)
  - The Differentiated Services architecture (DiffServ)
- QoS-related protocols:
  - RTP provides an adaptive multimedia transport protocol, that adjust to the properties of the underlying network
  - Multiprotocol Label Switching (MPLS) can be used to create dedicated links in a packet-switched network and provide a QoS-aware service

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Two Architectures: IntServ

- The IntServ model includes two sorts of services targeted towards real-time traffic: guaranteed and predictive service
- Resource reservation and admission control are key issues
- The term "guarantee" is used in a broad sense:
  - Absolute or statistical,
  - Strict or approximate
- Source describes its desired flow rate and sends this information to the routers and the receiver
- Network admits requests and reserves resources
- Source must send at this rate (controlled by network)
- Provides a sort of “dedicated” connection within an IP packet-switched network
- Reservation of resources are usually done with the Resource Reservation Protocol (RSVP)

IntServ: Guaranteed Service (GS)

- Aims to control the maximal queuing delay
- A flow is described using a Traffic Specification (TSpec): average and burst rates, average and max packet size, etc.
- Routers compute various parameters describing how it will handle the flow's data
- By combining the parameters from the routers in a path, it is possible to compute the maximum delay a packet will experience when transmitted via that path

IntServ: Controlled-Load Service (CL)

- Goals of the service:
  - High percentage of packets successfully delivered to the receiving end-nodes
  - The transit delay experienced by a high percentage of the packets will not greatly exceed the minimum transmit delay experienced by any delivered packet
- Clients requesting controlled-load service provide a TSpec of the data traffic they will generate
- The controlled-load service does not use specific parameters such as delay or loss
- An accepted request for CL service implies a commitment by the network to provide service closely equivalent to *uncontrolled (best-effort) traffic under lightly loaded conditions*

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Resource Reservation Protocol (RSVP)

- Used by a host to request specific qualities of service from the network routers for application packet flows
- Makes resource reservations for both unicast and multicast applications, adapts to group memberships and routes
- Receiver-oriented: receiver of a flow initiates and maintains the resource reservation used for that flow
- Sender of the data flow still informs the receiver about the traffic characteristics, to allow for a proper reservation from the network
- RSVP maintains soft state in routers and hosts; reservations can be modified and are not stored "forever" in routers

Resource Reservation Protocol (RSVP)

- RSVP makes reservations for unidirectional data flows
- It does not transport application data
- The traffic control and policy control parameters are opaque to RSVP
- RSVP provides several reservation models or "styles" to fit a variety of applications (shared and dedicated reservations)
- RSVP provides transparent operation through routers that do not support it: RSVP packets are just normal IP packets
- RSVP is not a routing protocol but depends upon present and future routing protocols
- RSVP supports both IPv4 and IPv6

IntServ & RSVP: Issues

- "All or nothing" -service
- Provides near absolute guarantees of QoS if the request is accepted (unless routers crash, mobile nodes move…)
- Need for state information, processing overhead and (sometimes heavy) signalling are the major concerns
- To be useful, all nodes on the path should support RSVP
- RSVP is also very complex because of the support for multicasting; today unicast would be mostly enough
- RSVP signalling in mobile networks is difficult
- RSVP, and other QoS protocols, have been analyzed by the IETF NSIS working group (RFC will be published very soon)

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Two Architectures: DiffServ

- No signalling, no reservations
- No feedback about the resource availability
- Traffic entering a network is marked with a code to differentiate a certain packet/flow from others
- Uses the old 8-bit IP TOS-field, 6 bits (2 bits used by ECN)
- Different codes result in different service at routers
  - Premium service, priority-based service, best-effort
- Does not provide absolute guarantees to flows
- Scales well to large numbers of users
- “All, or something, maybe” -service

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Two Architectures: DiffServ (cont.)

- Architecture for implementing service classes in the Internet, usually controlled by agreements between ISPs
- Architecture is composed
  - Per-hop forwarding behaviours (=classes),
  - Packet classification functions,
  - Traffic conditioning functions including metering, marking, shaping, policing, and
  - Service Level Agreements (SLA) between data senders and the forwarding network operator
- Per-Hop Behaviours (PHB) define the DS field handling (per link or “hop”) and service outcome of classes marked using the DS field in the IP header

Two Architectures: DiffServ (cont.)

- Complex classification and conditioning functions only at network boundary nodes (and possibly at sending host)
- Service differentiation separately per direction
- Two types of services standardized, operators may define other services and code points
- Does not necessarily provide end-to-end QoS:
  - Operators may have different meanings and implementations for classes and code points,
  - The code points can change, thus, may not remain the same on the whole end-to-end path.

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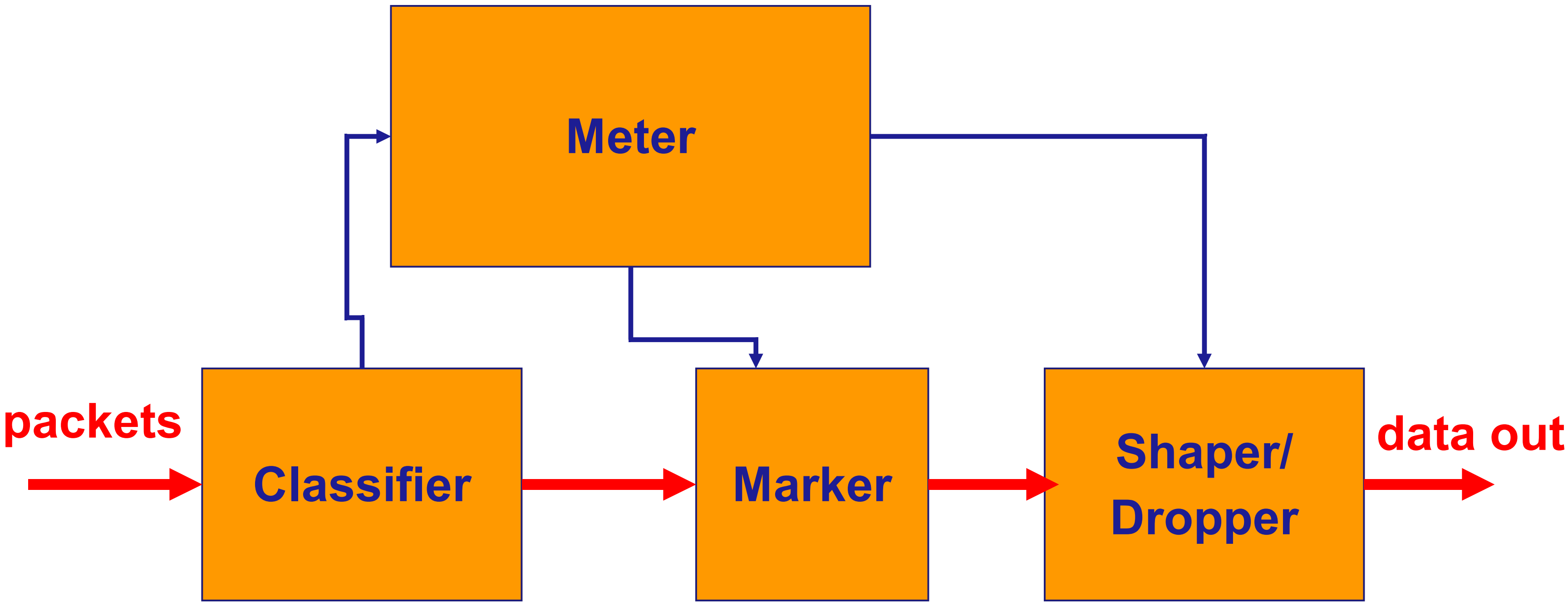
Two Architectures: DiffServ (cont.)



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DiffServ: Expedited Forwarding (EF)

- ”A Premium service”
- To build a low loss, low latency, low jitter, assured bandwidth, end-to-end service through DiffServ domains
- Mimics "virtual leased line”
- The aggregate has a well-defined minimum departure rate
- The standard does not define the implementation, just the nature of service and how to achieve it

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DiffServ: Assured Forwarding (AF)

- A “classic” priority-based service
- The AF PHB group provides four independent classes
- Each class has a certain share of the forwarding resources
- Within each AF class, an IP packet can be assigned one of three different levels of drop precedence
- A DiffServ node does not reorder IP packets of the same flow if they belong to the same AF class
- In case of congestion, the drop precedence of a packet determines the relative importance of the packet within the AF class

Stateless vs. Stateful Operation

- In IntServ the network must maintain the state of flows
- In DiffServ there is no need for states, the DSCP is enough
- Also the application does not give information of its traffic
- IntServ gives a fine level of granularity, while DiffServ is more approximate in its service outcome
- In IntServ, scalability and router performance are of concern
- DiffServ does not inform the client of the outcome of the service, applications need to monitor the service
- Aggregate RSVP, RSVP DCLASS Object, IntServ over DiffServ, and RSVP proxies are latest efforts to find better solutions

A Look into the Future

- The challenge is addressing the weaknesses of the two main architectures and integrate them (RFC2998):
  - IntServ/RSVP would provide the service signalling
  - DiffServ would be used for the core
  - RSVP allows applications to signal their needs to the network
  - RSVP requests are mapped to DSCPs for access admission and forwarding within the network
- The Next Steps in Signalling (NSIS) IETF WG is looking at new ways to do QoS signalling (ready soon)
- Another approach is to make applications and transport protocols behave better (e.g. DCCP)
- Lately, people have raised issues about TCP and "friendliness"
- A fundamental question remains:
  - What is the business model for QoS?
  - Who/when/why/how pays for a differentiated service?

Pointers

- IETF Web site: http://www.ietf.org/ (see working groups IntServ, DiffServ, RSVP, ISSLL, AVT (for RTP), NSIS, DCCP)
  - IntServ and RSVP: RFCs 2205, 2210, 2211, 2212, 2961
  - DiffServ: RFC 2474, 2475, 2597, 3246, 3247, 3248
  - IntServ and DiffServ together: 2990, 2998
  - NSIS: 4080, 4094,
  - RTP: RFC 3550
  - QoS Routing: RFC 2386
- RFCs at http://www.ietf.org/rfc/rfcXXXX.txt
- Also for RTP: http://www.cs.columbia.edu/~hgs/rtp/
- Very good book is (a little outdated, though):
  Geoff Huston: Internet Performance Survival Guide, QoS Strategies for Multiservice Networks, Wiley, 2000.

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