Quality of Service for Packet Networks

Real-Time and Embedded Systems (M)

Lecture 17

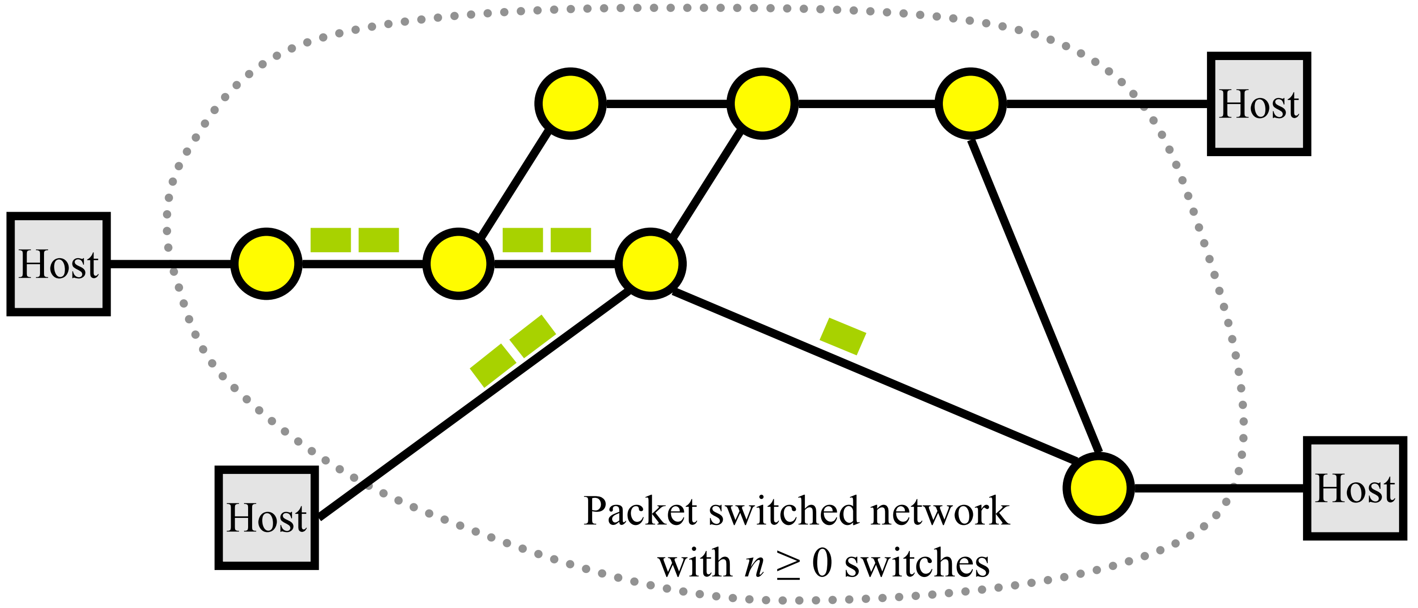
UNIVERSITY *of* GLASGOW

Lecture Outline

- Best effort versus enhanced services
- Queuing disciplines
  - Weighted fair queuing and variants
  - Weighted round robin
- Resource reservation protocols
  - RSVP

- Material corresponds to parts of chapters 7 and 11 of Liu’s book

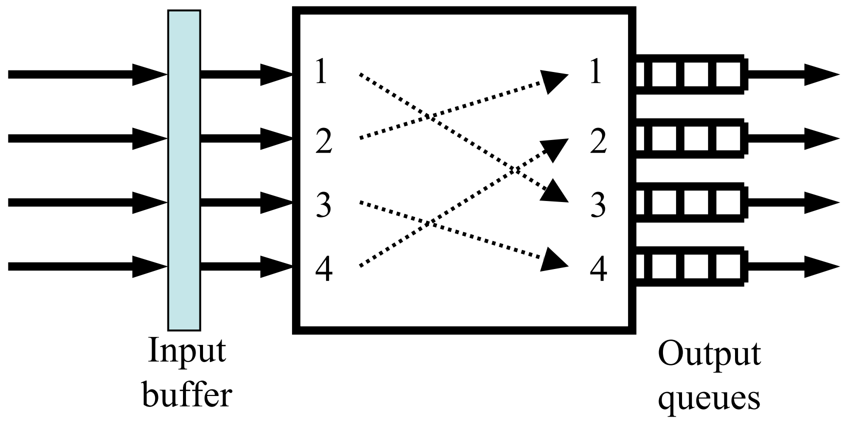
Model of Packet Switched Networks



Links have constant *propagation delay*

Switches queue packets for transmission if output link busy (additional variable delay)

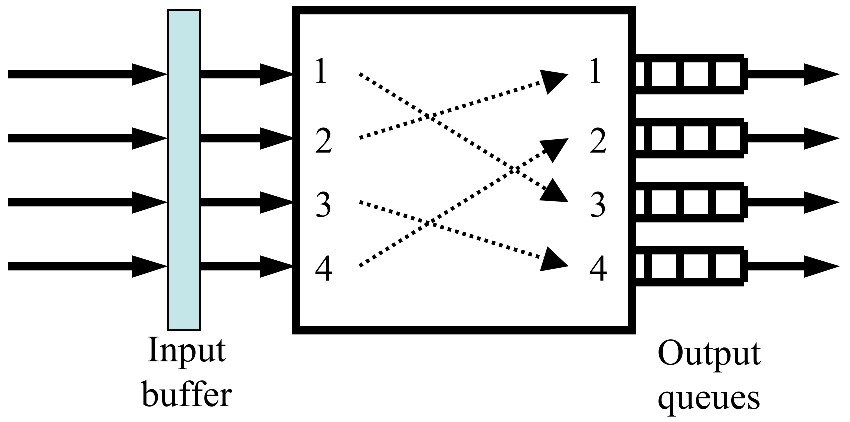
Choice of *job scheduling algorithm* on the output link is critical for real time traffic



Best Effort versus Enhanced Service

- Best effort networks use a single output queue for each link
  - FIFO with drop tail
  - FIFO with random drop (RED)

  and don’t control the output queuing
- Uncontrolled best effort networks are inexpensive, but don’t provide rate guarantees or control the jitter
- Enhanced service packet networks provide this control, and are better suited to real-time traffic
  - Packets in the output queues are scheduled for transmission to affect some policy, rather than in FIFO order



How to Implement Enhanced Service?

- To schedule packets according to some policy, policy must be communicated to the network, and the network must perform admission control to ensure that policy constraints can be met
- Implies the network must implement:
  - A packet scheduling algorithm
    - To prioritise certain classes of traffic
    - To manage the output queues
  - Admission control
    - To determine if the signalled flows can be supported
  - A signalling protocol
    - To communicate the stream characteristics to the network
      - Flow specification
      - Required performance

Service Disciplines for Enhanced Services

- The combination of scheduling algorithm and acceptance test is a *service discipline*
- Used to control jitter and packet rate
  - Ensure flows receive their *proportional fair share* of capacity
    - Rates controlled to allocate capacity proportionally, according to policy
    - Algorithms can be *rate allocating* or *rate controlled*
      - Rate controlled algorithms give each flow an allocated rate, and never let flows exceed their rate
      - Rate allocating algorithms give each flow an allocated rate, but let flows exceed their rate if there is spare capacity
    - Flows serviced regularly, to avoid starvation
  - Ensure *timing isolation* between flows
    - Partly as a side-effect of rate control
    - Some algorithms perform explicit jitter control, preserving the traffic pattern – inter-packet spacing – when forwarding traffic

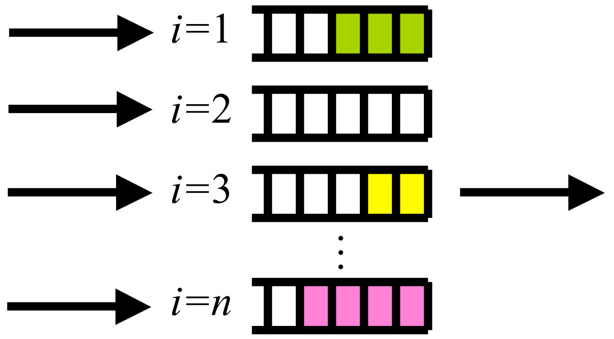
Priority Queuing Algorithms

- Two priority packet scheduling algorithms widely implemented:
  - Weighted round robin (WRR)
  - Weighted fair queuing (WFQ)

[See also lecture 8]

Weighted Round Robin Scheduling

- In *round robin* scheduling, jobs are placed in a FIFO queue
  - The job at the head of the queue executes for one time slice
  - If it doesn't complete within the time slice, it is pre-empted and put at the back of the queue
  - There are *n* jobs in the queue, each job gets one slice every *n* time slots (that is, every *round*)
- A *weighted round robin* schedule extends this, to give each job *i* a weight wt_i
  - A job with weight wt_i executes for wt_i time slices each round
  - Length of the round equals $\sum_{i=1}^{n} wt_i$



Set of *n* flows destined for an output link

wt_i packets from queue *i* sent in turn

WRR Scheduling: Throughput Guarantees

- Assume constant bit rate, periodic, flows: $M_i = (p_i, e_i, D_i)$
  - Minimum inter-arrival time of messages p_i
  - Size of each message e_i
  - Maximum acceptable end-to-end delay D_i

- Each round, if more than wt_i packets are backlogged on queue i, then wt_i packets are transmitted
  - Each flow is guaranteed wt_i slots each round
  - Rate allocating: may send more, if nothing else to transmit
- A design parameter is RL the maximum number of slots per round
  - At all times $\sum_{i=1}^{n} wt_i \leq RL$
  - Each flow is guaranteed a share wt_i/RL of the link capacity
  - Provided that:
    - $RL < p_{\text{min}}$ (where p_{min} is minimum p_i over all i)
    - $wt_i \geq e_i/(p_i/RL)$ (with appropriate rounding)

WRR Scheduling: End-to-End Delay Bound

- Messages take at most e_i/wt_i rounds to complete
- Implies delay through first switch = $(e_i/wt_i)RL$
- At each subsequent switch, each round of packets arriving is sent in the next round
  - Implies one round delay at each hop

- Therefore, end-to-end delay for connection i with message size e_i assigned weigth wt_i passing through r switches is bounded by:

$$W_i \leq (e_i/wt_i + r - 1)RL$$

- Can also be shown that jitter can be bounded by

$$\text{jitter} < p_i - e_i + (r - 1)(RL - 1)$$

for messages of size e_i with inter-arrival time of p_i

WRR Scheduling: Connection Setup

- Why use a fixed round length *RL*?
- Too costly to change the round length each time a new flow is established
  - Would require adjusting weights for all pre-existing flows

- With a fixed *RL*, connection establishment becomes:
  - Pass parameters (p_i, e_i, D_i) to each hop router
  - At each hop, the scheduler computes the weight, wt_i, required to support the new flow
  - If the sum of existing weights < RL - wt_i the flow is accepted at that hop
  - If all hops accept, the flow is established

Weighted Round Robin Scheduling

- Flows are guaranteed capacity
- WRR scheduling is efficient to implement, since the scheduling decision is O(1)
  - Simply pick wt_i packets from the next queue

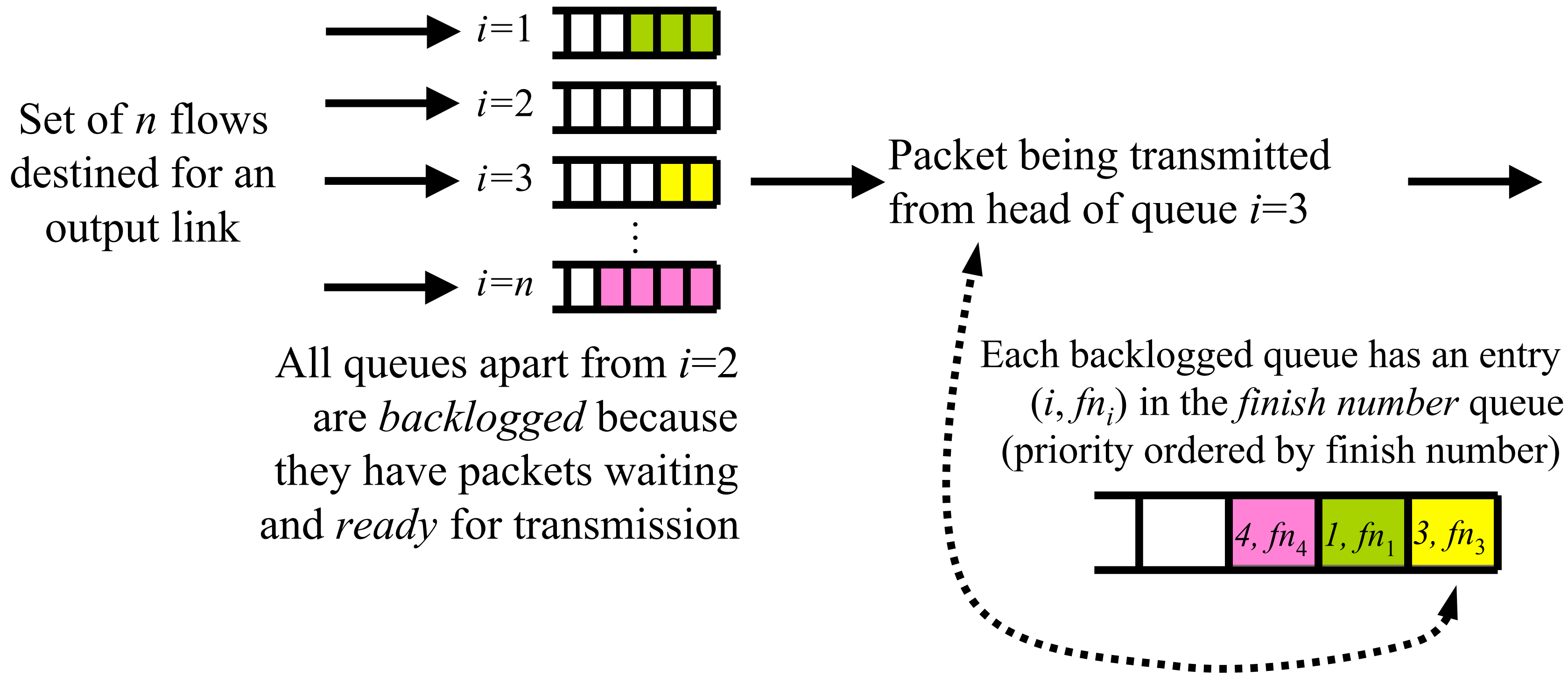
- End-to-end delay can be bounded
- Since the scheduler is rate allocating, jitter is not controlled but can be bounded

Weighted Fair Queuing

- "Packet-by-packet generalized processor-sharing algorithm"
  - A rate allocating service discipline; provides each flow with at least its proportional fair share of link capacity; isolates timing between flows
- Definitions:
  - A packet switch has several inputs, feeding to an output link shared by *n* established flows
    - Each flow, *i*, is allocated a fraction $\tilde{u}_i$ of the link
    - Total bandwidth allocated to all *n* connections is $U = \sum_{i=1}^{n} \tilde{u}_i$ where $U \leq 1$
    - Assume an acceptance test rejects connections that would cause requested bandwidth to exceed available bandwidth
  - Define the "finish number", fn_i, to represent job completion times
    - Used in definition of scheduling algorithm

WFQ: Packet Scheduling

- Each link of the packet switch is output buffered
- Output buffers conceptually comprise two sets of queues:
  - A set of FIFO queues for each of the *n* flows
  - A priority ordered shortest finish number (SFN) queue
- Entry at head of SFN queue indicates the FIFO queue to service



Set of *n* flows destined for an output link
$i=1$
$i=2$
$i=3$
$i=n$
All queues apart from $i=2$ are *backlogged* because they have packets waiting and *ready* for transmission
Packet being transmitted from head of queue $i=3$
Each backlogged queue has an entry (i, fn_i) in the *finish number* queue (priority ordered by finish number)
$4, fn_4$
$1, fn_1$
$3, fn_3$

WFQ: Packet Scheduling

- As a packet becomes ready on a FIFO queue, its finish number is calculated, and the SFN queue is updated
  - Currently transmitting packet never pre-empted, even if the finish number of the newly ready packet would place it at the head of the SFN queue
- When a packet completes transmission, it is removed from the head of the FIFO and SFN queues
  - If the FIFO queue is still backlogged, the SFN queue is updated with the finish number of the newly ready packet
  - The packet from the queue referenced by the entry at the head of the SFN queue begins transmission

- Key is the calculation of the finish number for each packet as it becomes ready on a backlogged queue

WFQ: Finish Numbers

- Define:
  - The total bandwidth of all backlogged flows, U_b
  - The finish number of the link, FN
  - The current time, t, and the previous time, t_{-1}, when FN and U_b updated

- Computing the finish number when the link becomes active:
  - The link is idle: FN=0, U_b=0, t_{-1}=0 and all $fn_i = 0$
  - A packet of length e arrives on a flow assigned fraction $\tilde{u}_i$ of the link, and starts a link busy interval on link i
    - Compute $U_b = U_b + \tilde{u}_i$ and $fn_i = fn_i + e/\tilde{u}_i$
    - Set $t_{-1} = t$
    - Insert entry (fn_i, i) in the SFN queue
  - Intuition: finish number of the first packet set to transmission delay for the job, adjusted by the fraction of the link used

[cont'd]

WFQ: Finish Numbers

- Computing subsequent finish numbers during link busy interval
  - If a packet arrives on a previously idle flow, i
    - Increment FN by $(t - t_{-1})/U_b$
    - Compute $fn_i = \max(FN, fn_i) + e/\tilde{u}_i$
    - Insert entry (fn_i, i) in the SFN queue
    - Set $t_{-1} = t$ and increment $U_b = U_b + \tilde{u}_i$
  - When the transmission of a packet on flow i completes
    - If the connection remains backlogged
      - Compute $fn_i = fn_i + \mathrm{e}/\tilde{u}_i$ where e is the length of the newly ready packet
      - Insert entry (fn_i, i) in the SFN queue
    - If the connection becomes idle
      - Increment FN by $(t\text{-}t_{-1})/U_b$
      - Set $t_{-1} = t$ and decrement $U_b = U_b - \tilde{u}_i$
  - Intuition: finish number fn_i represents deadline when a packet on flow i will be transmitted

WFQ: Properties

- Complex algorithm to calculate finish number, and determine the transmission order of packet – what is the benefit?

- Can control latency and jitter, can isolate traffic flows
  - Bounds on per-hop and end-to-end latency for traffic
  - Guaranteed network capacity

WFQ: Per-Hop Latency

- Delay between time a packet becomes ready (when it reaches the head of the FIFO output queue) and when transmission completes is *latency*, L_i
  - Blocking time due to the WFQ algorithm itself, ignoring queuing delay

- It has been proved that $L_i < e_i/\tilde{u}_i + 1$
  where: e_i is the normalised maximum packet length,
  $\tilde{u}_i$ is the fraction of the link assigned to this flow
  - First term: time taken to transmit largest packet
  - Second term: blocking due to non pre-emptive schedule

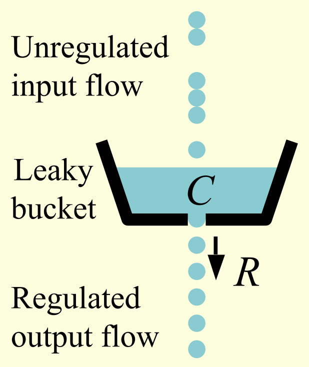
- Because of the rate control behaviour of WFQ, this bound is independent of other traffic on the output link

WFQ: Total Per-Hop Delay

- Total per-hop delay, $W_i(1)$, for a packet of length e is equal to the sum of latency, calculated previously, and queuing delay
- To predict queuing delay, you need to know arrival pattern
  - Queuing delay can be unbounded even if allocated bandwidth, $\tilde{u}_i$, equals the actual bandwidth of the flow, u_i, if no constraint on arrivals
  - But, can be proven that $W_i(1) = (E_i + e_i)/\ \tilde{u}_i + 1$ if arrivals fit a (u_i, E_i) leaky bucket constraint and flow allocated sufficient fraction $\tilde{u}_i \geq u_i$ of link
    - (Latency term) + E_i to represent queuing delay
    - Matches periodic, and many sporadic, isochronous flows

Capacity

Leak rate

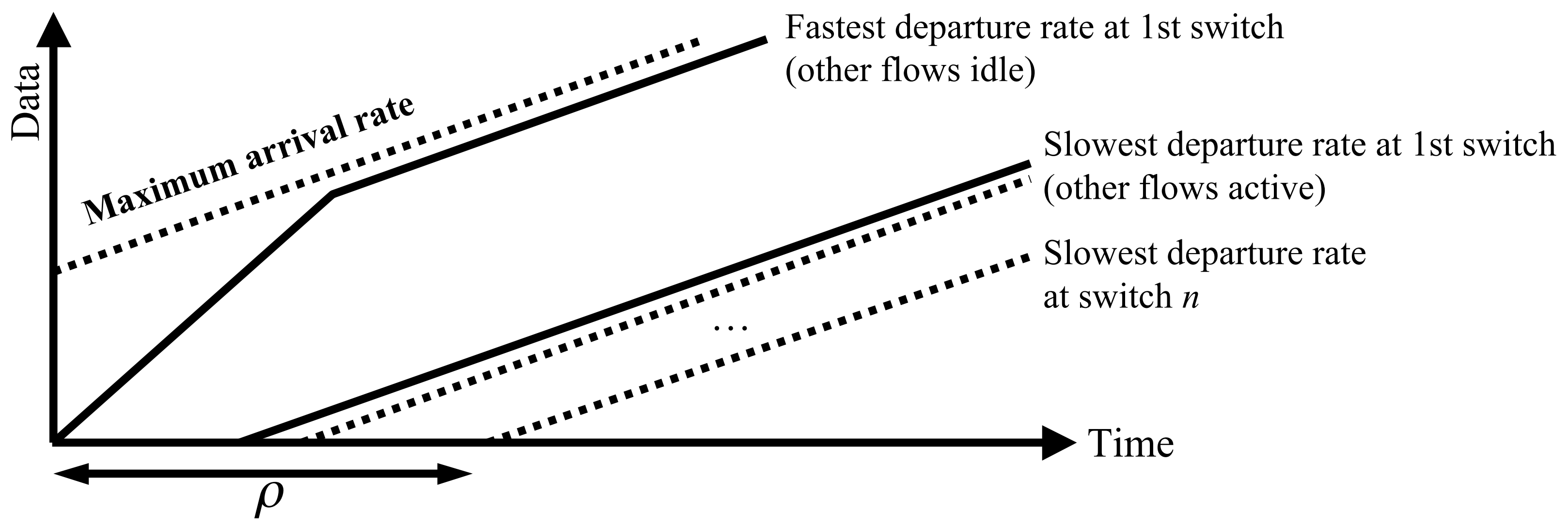


- Output data rate will not exceed R, irrespective of variation in the input rate (may be $< R$, if input is idle)
- Average input rate cannot exceed R, except for short term bursts of at most C packets; maximum number of packets entering the network in any given time, t, is $Rt + C$
- Useful for turning sporadic flows into periodic flows, if average rate of sporadic flow $\leq R$, and bursts never cause bucket overflow

End-to-End Delay of WFQ

- If we know per-hop delay, can we model end-to-end delay?
- Assume a homogeneous network:
  - A connection i with rate u_i traverses ρ switches
  - Traffic is initially shaped to match a (u_i, E_i) leaky bucket
  - Intermediate switches perform WFQ, but no traffic shaping
  - All links have the same capacity, and the connection is allocated the same fraction $\tilde{u}_i = u_i$ of bandwidth

End-to-End Delay of WFQ


Data
Maximum arrival rate
Fastest departure rate at 1st switch (other flows idle)
Slowest departure rate at 1st switch (other flows active)
Slowest departure rate at switch *n*
…
Time
ρ

- Making worst case assumptions, maximum arrival rate at switch *n* is slowest departure rate at switch *n*-1
  - 1 unit of delay added due to non pre-emption at each hop

- Can derive $W_i(\rho) = \frac{E_i + \rho e}{\tilde{u}_i} + \rho$ when $\tilde{u}_i = u_i$

  (Same as per-hop delay, but adjusted for the number of hops *ρ)*

End-to-End Delay of WFQ

- Generalise: output links may have different transmission rates
- For switch *j* traversed by flow *i*
  - Assume flow *i* satisfies a leaky bucket (λ_i, E_i) at the first hop
  - Assume flow *i* is allocated $u_i = \lambda_i$
  - Let $e_{\max}(i, j)$ denote time taken to transmit largest packet of all flows sharing the output link with connection *i*

- Can show that $W_i(\rho) = \dfrac{E_i + \rho e}{\lambda_i} + \sum_{j=1}^{\rho} e_{\max}(i, j)$
  - As before, but adjusted for non pre-emption delays of variable rate/size packets at each hop
- Can also show that, maximum jitter is $\dfrac{E_i}{\lambda_i} + \sum_{j=i}^{\rho} e_{\max}(i, j)$

Weighted Fair Queuing: Summary

- A dynamic priority scheduling algorithms to ensure:
  - Each flow *i* gets at least a fraction u_i of the link bandwidth
  - Packets are scheduled fairly, and starvation is avoided
- Per-hop delay and end-to-end delay for a flow can be bounded, if the traffic pattern of the flow is known
  - Independent of the other flows in the network
- Compared to an uncontrolled packet network WFQ is complex, but can guarantee throughput, latency and jitter
  - Simplifies applications running on a WFQ network, since they can predict timing of message delivery

Resource Reservation Protocols

- Throughout the discussion of queuing algorithms, we have assumed that the required rate allocation, $\tilde{u}_i$, is known at each switch
- In a real packet network, hosts must inform the routers of the flow characteristics and required rate
- Implies a *resource reservation protocol* is needed
- Several issues to consider:
  - Scalability and router state
  - Multicast communication
  - Heterogeneity of destinations
  - Dynamic membership
  - Relation to routing and admission control

Case Study: RSVP

- A standard resource reservation protocol in the Internet is RSVP
- Basic operation:
  - Sources send periodic *path* messages, describing the flow
    - Create path state in intermediate routers
  - Receivers send *reservation* messages back towards the source
    - Cause intermediate routers to perform acceptance test and setup a resource reservation for the flow described by the path messages
    - May send a reject message to the receiver, if acceptance test fails
    - Reservations refreshed periodically by receivers
- Characteristics:
  - Soft state, for graceful failure
  - Receiver driven reservations support multicast

- Widely supported and available if you control the network, but not widely used in the public Internet

Summary

- Why enhanced service is needed
- What is needed to support enhanced services
  - Queue discipline
  - Acceptance test
  - Signalling protocol
- Two approaches to implementing priority queuing
  - WFQ
  - WRR
  - Performance trade-off between the two approaches
- Brief pointer to RSVP