10.2 Weighted Fair Queuing (WFQ)

- Configure the point-to-point Serial link between R4 and R5 with an interface clock rate and interface bandwidth of 128Kbps.
- Set the output hold-queue size to 256.
- Configure WFQ on the links with a length of 16 for the congestive discard threshold, 128 for the number of conversations, and 8 for RSVP reservable queues.
- Set the tx-ring size to the minimal value to engage the WFQ as fast as possible.
- Normalize the packet flows for the link by ajdusting the MTU so that each IP packet takes no more than 10ms to be sent.

Configuration

```
R4:
interface Serial0/1
 clock rate 128000
 bandwidth 128
 tx-ring-limit 1
 fair-queue 16 128 8
 hold-queue 256 out
 ip mtu 156

R5:
interface Serial0/1
clock rate 128000
 bandwidth 128
 tx-ring-limit 1
 fair-queue 16 128 8
 hold-queue 256 out
 ip mtu 156
```

Verification

> ✎ **Note**
>
> WFQ uses an intelligent congestion management solution that provides "fair" sharing of the interface bandwidth between multiple traffic *flows*. A traffic "flow" (or *conversation*) is a unidirectional sequence of packets, defined based on the protocol type, the source/destination IP addresses, the source/destination ports numbers (when available), and partially on the IPv4 ToS byte value. For example, an HTTP file transfer between two hosts represents one packet flow, while ICMP packets sent from one host to another represents a second.

The term "fair" on WFQ refers to the *max-min* fairness. The WFQ calculation procedure is as follows.

First, divide the interface bandwidth equally between all flows. For example if there are 10 flows, and 100Kbps of bandwidth, each flow gets 10Kbps. If a flow demand is less than the "equal" share, e.g. a flow only needs 5Kbps instead of 10Kbps, allocate the unneeded bandwidth equally among the remaining flows. If there are flows that demand more than the equal share, e.g. the equal share is 10Kbps, but two flows demand 25Kbps and 20Kbps respectively, they will each get the equal, maximum, possible shares (e.g 19Kbps and 19Kbps) but only if there are flows which demand less than the equal share.

In this context *max-min* means that all flows first get the bare minimum, but the procedure tries to maximize each flow's share if possible. The concept of basic WFQ is very important to understand as many other congestion management techniques utilize it, such as Round Robin scheduling. To reiterate, the key fact in WFQ is that the *max-min* scheduling allows the sharing of a flow's unclaimed bandwidth between other flows.

An individual flow in the queue may be more or less demanding than other flows. The "demanding" flow generates the higher bit-rate, either due to larger packet sizes or a higher packet per second rate. For example compare a bulk FTP file transfer against telnet sessions or VoIP RTP streams. Within the context of flow-based sharing, it is also important to understand that a single host may generate *multiple* flows, such as with with P2P file-sharing applications or download "accelerators", thus this particular host can obtain an "unfair" share of bandwidth compared to other hosts. This, unfortunately, is a limitation of a classification scheme that based on simple flows, such as WFQ, which has no notion of "flow mask".

To enhance its scheduling logic, WFQ may assign a *weight* value to a flow. The weight affects the minimum guaranteed share of bandwidth available to a flow. If there are N flows with weights w1, w2 ... wN, then flow K will get the minimum share of bandwidth where sK=(w1+w2+… wK+… wN)/wK – inversely proportional to its weight. The shares are relative, in the sense that the scheduler divides the available bandwidth in proportions: s1:s2:…sN.

IOS implementation of WFQ assigns weights automatically based on the IP Precedence (IPP) value in the packet's IP header. The formula is Weight=32384/(IPP+1), where IPP is the IP Precedence of the flow.

To understand the effect of the weight settings, imagine four flows with different IP Precedence values of zero, one, one and three.

Step 1:
Using the above formula for weight, we obtain:

Weight(1) = 32384/(0+1) = 32384
Weight(2) = 32384/(1+1) = 16192
Weight(3) = 32384/(1+1) = 16192
Weight(4) = 32384/(3+1) = 8096

Step 2:

Compute the sum of all weights:
Sum(Weight(i),1…4) = 32384+16192*2+8096 = 72864

Step 3:
Compute the shares:

Share(1) = 72864/Weight(1) = 72864/32384 = 2.25
Share(2) = 72864/Weight(2) = 72864/16192 = 4.5
Share(3) = 72864/Weight(3) = 72864/16192 = 4.5
Share(4) = 72864/Weight(4) = 72864/8096 = 9

The proportion is 2.25:4.5:4.5:9 = 1:2:2:4 – this is based on the "shifted" ip precedences: (IPP(i) + 1)

Note that the numerator "32384" is arbitrary to those computations, meaning that you can use any value. What is important however is the proportion of "shifted" IP precedences.

For bandwidth sharing to be correct, all flows must be adaptive. Adaptive means that a flow must respond to packet drops by slowing its sending rate. Non-adaptive, aggressive flows may defeat the bandwidth sharing logic of WFQ by claiming all available *buffer space* and starving other flows. This is directly linked to the fact that all flows use shared buffer space when implementing WFQ (more on this later).
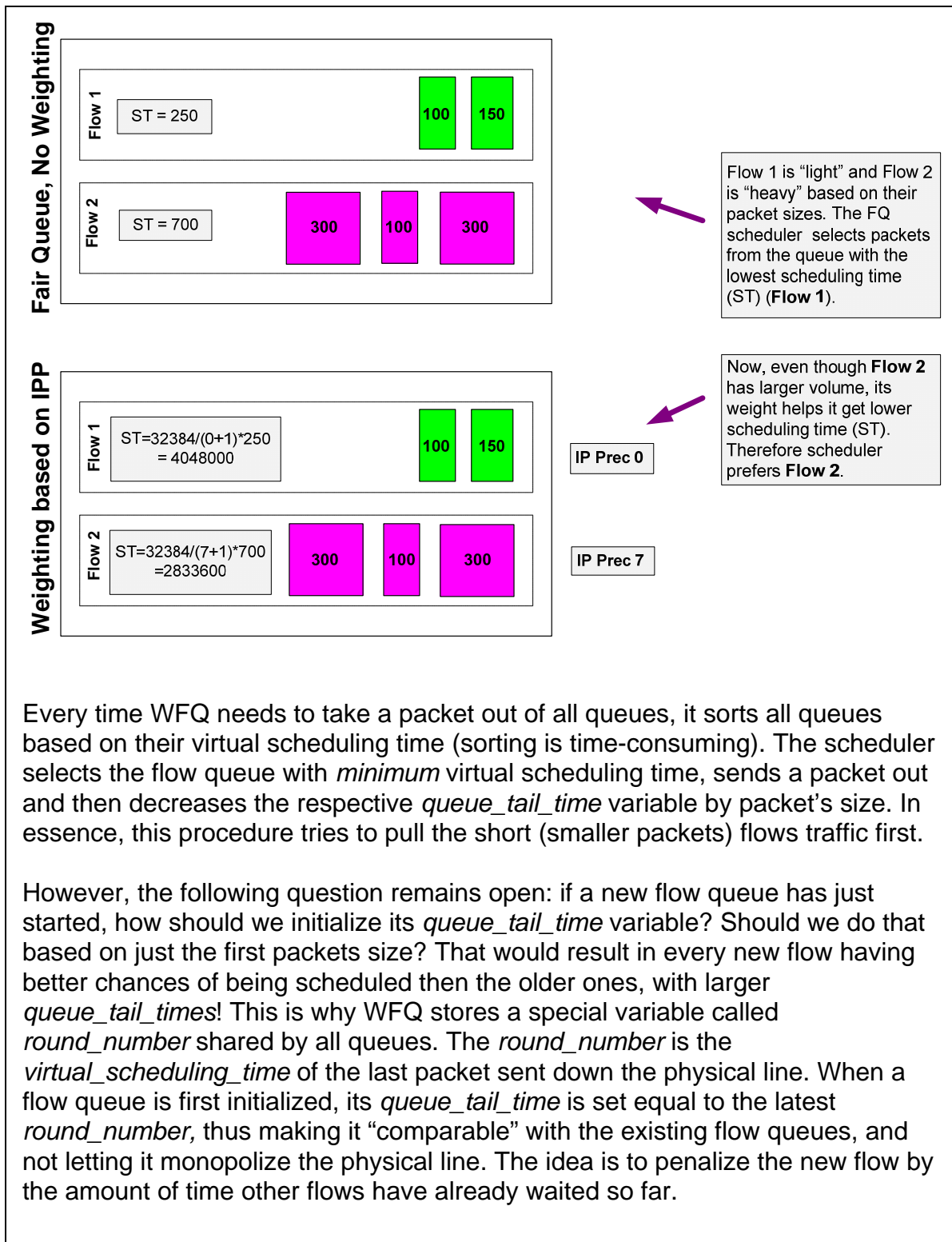
Specifically WFQ implements its fair sharing logic as follows. First, the scheduler creates a group of flow queues for the interface, e.g. 256 "conversations", based on a manual setting or an automatic calculation derived from the interface bandwidth. When a new packet arrives for output scheduling, the scheduler applies a special *hash* function to the packet's source/destination IP/Port values to yield the flow queue number. This is why the number of queues is always a power of 2, because the hash the output value is somewhere between 0 and 2^N. This procedure also means that multiple flows may share the same queue, called "hash collision", when the number of flows is large.

Each flow queue has a special *virtual scheduling time* assigned – the amount of "time" that would take to serialize the packets in the queue across the output interface. This "virtual time" is actually the total size of all packets stored in the flow queue scaled by the flow computational weight.

Now imagine a new packet of size *packet_size* and IP precedence *packet_ip_precedence* arrives to its respective flow queue (hash queue):

weight = 32384/(packet_ip_precedence +1)
*virtual_scheduling_time = queue_tail_time + packet_size*weight*
queue_tail_time = virtual_scheduling_time

The "*queue_tail_time*" variable stores the previous virtual scheduling time for the flow. Note that the weight is inversely proportional to IP precedence, and thus more important packets have smaller virtual scheduling time value (WFQ thinks that it can serialize them "faster"). It is more appropriate to call this computational weight the "scaling factor" to avoid confusion with the classic meaning of weight.

Fair Queue, No Weighting

Flow 1 | ST = 250 | 100 | 150

Flow 2 | ST = 700 | 300 | 100 | 300

Flow 1 is "light" and Flow 2 is "heavy" based on their packet sizes. The FQ scheduler selects packets from the queue with the lowest scheduling time (ST) (**Flow 1**).

Weighting based on IPP

Flow 1 | ST=32384/(0+1)*250 = 4048000 | 100 | 150 | IP Prec 0

Flow 2 | ST=32384/(7+1)*700 =2833600 | 300 | 100 | 300 | IP Prec 7

Now, even though **Flow 2** has larger volume, its weight helps it get lower scheduling time (ST). Therefore scheduler prefers **Flow 2**.

Every time WFQ needs to take a packet out of all queues, it sorts all queues based on their virtual scheduling time (sorting is time-consuming). The scheduler selects the flow queue with *minimum* virtual scheduling time, sends a packet out and then decreases the respective *queue_tail_time* variable by packet's size. In essence, this procedure tries to pull the short (smaller packets) flows traffic first.

However, the following question remains open: if a new flow queue has just started, how should we initialize its *queue_tail_time* variable? Should we do that based on just the first packets size? That would result in every new flow having better chances of being scheduled then the older ones, with larger *queue_tail_times*! This is why WFQ stores a special variable called *round_number* shared by all queues. The *round_number* is the *virtual_scheduling_time* of the last packet sent down the physical line. When a flow queue is first initialized, its *queue_tail_time* is set equal to the latest *round_number,* thus making it "comparable" with the existing flow queues, and not letting it monopolize the physical line. The idea is to penalize the new flow by the amount of time other flows have already waited so far.

The next important point is the congestive discard threshold (CDT), which is a unique congestion avoidance scheme available with WFQ. First you configure the total buffer space allocated to WFQ using the **hold-queue <N> out** command. The WFQ shares this buffer space between all flow queues. Any queue may grow up to the maximum free space, but as soon as its size reaches the CDT, WFQ drops a packet from a flow queue with the *maximum* virtual scheduling time (this may be some other queue, not the one that crossed the packet threshold). This way, WFQ penalizes the most aggressive flow and triggers a mechanism similar to random early detection's (RED) prevention of tail-drop. The fair-queue settings command is **fair-queue <CDT> <N Flow Queues> <N Reservable Queues>.**

The number of reservable conversations (queues) is the number of flow-queues available to RSVP reservations (if any). Those flows have a very small weight value, and thus are preferred above any other flows. In addition to reserved flow queues, there are special "Link Queues". The number of queues is fixed to 8, and they are numbered right after the maximum dynamic queue (e.g. if there are 32 dynamic queues, "Link Queues" start at number 33). WFQ uses those queues to service routing protocol traffic and Layer 2 keepalives – everything that is critical to router operations and management. Each queue has a weight 1024, which is lower than any dynamic queue can get, so control plane traffic has priority over regular data traffic.

Finally, note that the interface bandwidth setting does not influence the WFQ algorithm directly. It only prompts the WFQ command to pick up optimal parameters matching the interface bandwidth. The bandwidth, however, *is* used for admission control with RSVP flows.

For this particular task, verification can be performed as follows.

```
Rack1R4#show queueing interface serial 0/1
Interface Serial0/1 queueing strategy: fair
  Input queue: 0/75/0/0 (size/max/drops/flushes); Total output drops: 2044
  Queueing strategy: weighted fair
  Output queue: 3/256/16/2044 (size/max total/threshold/drops)
     Conversations  2/3/32 (active/max active/max total)
     Reserved Conversations 0/0 (allocated/max allocated)
     Available Bandwidth 96 kilobits/sec

Rack1R4#show queueing fair
Current fair queue configuration:

  Interface          Discard   Dynamic  Reserved  Link    Priority
                     threshold queues   queues    queues  queues
  Serial0/0          64        256      0         8       1
  Serial0/1          64        32       1         8       1
```

Changing the `ip mtu` for the interface in this task causes fragmentation at layer 3, and normalizes the packet sizes for further testing. With an additional 4 bytes of overhead for the layer 2 HDLC encapsulation, every frame sent out the link has a maximum size of 160 bytes (156 of IP plus 4 bytes of HDLC) . With a physical clocking rate of 128000 bits per second, this means a 160 byte packet will take a maximum time of 10ms to serialize. Adjusting serialization delay through fragmentation is covered in more detail later in this document.

To verify how WFQ shares the interface bandwidth, configure the network as follows:

1) Enable HTTP servers on R6 and R1 and set the root directory to "flash:".

2) Shutdown the Frame-Relay interface of R5 to ensure traffic from VLAN146 takes the path across the serial link between R4 and R5.

3) Configure R6 to mark traffic leaving the VLAN146 interface with an IP precedence of 1, and configure R1 to mark traffic leaving VLAN146 with an IP precedence of 3.

4) Create a policy-map on R5 to match incoming IP precedence 1 and IP precedence 3 packets, and apply it inbound to interface Serial 0/1. We will use this policy to meter incoming traffic.

5) Copy IOS images stored in the flash memory of R1 and R6 to SW2 and SW4's "null:" destinations using HTTP.

The resulting configuration is as follows.

```
R1:
ip http server
ip http path flash:
!
policy-map MARK
 class class-default
  set ip precedence 3
!
interface FastEthernet 0/0
 service-policy output MARK

R5:
class-map match-all IP_PREC3
  match ip precedence 3
 class-map match-all IP_PREC1
  match ip precedence 1
!
 policy-map METER
  class IP_PREC1
  class IP_PREC3
!
interface Serial 0/1
 service-policy METER input
 load-interval 30
 clock rate 128000
!
interface Serial 0/0
 shutdown

R6:
ip http server
ip http path flash:
!
policy-map MARK
 class class-default
  set ip precedence 1
!
interface FastEthernet 0/0.146
 service-policy output MARK
```

Rack1SW2#copy http://admin:cisco@155.1.146.6/c2600-adventerprisek9-mz.124-10.bin null:

Rack1SW4#copy http://admin:cisco@155.1.146.1/c2600-adventerprisek9-mz.124-10.bin null:

The configuration effectively generates two TCP flows across R4's connection to R5. Since TCP is adaptive, the flows should eventually balance and start sharing bandwidth using their weights: 1+1 and 3+1, which is 2:4. This means file transfers from R1 to SW4 should have twice as much bandwidth as the file transfer from R6 to SW2.

```
Rack1R5#show policy-map int serial 0/1
 Serial0/1

  Service-policy input: METER

    Class-map: IP_PREC1 (match-all)
      91618 packets, 17299811 bytes
      30 second offered rate 41000 bps
      Match: ip precedence 1

    Class-map: IP_PREC3 (match-all)
      330231 packets, 75523458 bytes
      30 second offered rate 83000 bps
      Match: ip precedence 3

    Class-map: class-default (match-any)
      58353 packets, 8839968 bytes
      30 second offered rate 0 bps, drop rate 0 bps
      Match: any
```

Check the flow queues on R4 to see their WFQ parameters. Note that the average packet length is the same, and the inverse weight proportion is 1/8096:1/16192 = 2:1 (the guaranteed shares of bandwidth for IP Precedence 1 and IP Precedence 0 traffic).

```
Rack1R4#show queueing interface serial 0/1
<snip>
  (depth/weight/total drops/no-buffer drops/interleaves) 7/16192/0/0/0
  Conversation 20, linktype: ip, length: 580
  source: 155.1.146.6, destination: 155.1.58.8, id: 0x6927, ttl: 254,
  TOS: 32 prot: 6, source port 80, destination port 11009

  (depth/weight/total drops/no-buffer drops/interleaves) 6/8096/0/0/0
  Conversation 26, linktype: ip, length: 580
  source: 155.1.146.1, destination: 155.1.108.10, id: 0x7BCF, ttl: 254,
  TOS: 96 prot: 6, source port 80, destination port 11001
```

As mentioned above, non-adaptive flows, such as UDP/ICMP traffic floods, may oversubscribe the shared WFQ buffer space. This is due to their "greedy" behavior - a single flow tries to monopolize the whole buffers space available to all queues, causing excessive packet drops, and dose not respond to congestive discards.

To see how this behavior can manifest itself originate two ICMP packet floods from R6 and R1 towards R5 using a packet size of 100 and a timeout of 0. The timeout value of zero means that the devices do not wait for an echo-reply before sending its next echo. Additionally configure R5 so that it does not respond to the echos so the return traffic doesn't affect the flow.

```
R5:
access-list 100 deny icmp any host 155.1.45.5
access-list 100 permit ip any any
!
interface Serial 0/1
 ip access-group 100 in
 no ip unreachables
```

Rack1R1#ping 155.1.45.5 repeat 100000000 timeout 0
Rack1R6#ping 155.1.45.5 repeat 100000000 timeout 0

Rack1R4#show queueing interface serial 0/1
```
<snip>
  (depth/weight/total drops/no-buffer drops/interleaves) 43/8096/7289/0/0
  Conversation 30, linktype: ip, length: 104
  source: 155.1.146.1, destination: 155.1.45.5, id: 0x05E1, ttl: 254, prot: 1

  (depth/weight/total drops/no-buffer drops/interleaves) 21/16192/25458/0/0
  Conversation 3, linktype: ip, length: 104
  source: 155.1.146.6, destination: 155.1.45.5, id: 0xC38A, ttl: 254, prot: 1
```

Rack1R5#show policy-map interface serial 0/1
```
 Serial0/1

  Service-policy input: METER

    Class-map: IP_PREC1 (match-all)
      104422 packets, 22233791 bytes
      30 second offered rate 47000 bps
      Match: ip precedence 1

    Class-map: IP_PREC3 (match-all)
      353680 packets, 85227338 bytes
      30 second offered rate 68000 bps
      Match: ip precedence 3
<snip>
```

Note the huge number of drops due to the queue getting full. Also, note that offered rates on R5 are *not* in a 1:2 proportion, even though the weights are set to share the bandwidth in 1:2 ratio and the packet lengths are the same.