Quality of Service – QoS Udemy Course Notes

Definition – providing a better quality to certain applications at the detriment of other applications.

Managed Unfairness

Priority to some sessions over other sessions

Voice and Video (Usually UDP) (Delay and Drop Sensitive) get priority over non delay sensitive applications

Three Traffic Types

* Data
* Voice
* Videos

**Data**

Bursty

Greedy – grabs as much bandwidth as is allocated

Drop-insensitive – if a packet is dropped, it is just resent

Delay-insensitive – latency is not a problem

TCP – most common for data applications, as there is guaranteed delivery

2 major types of Data applications

Interactive and Non-interactive

Telnet (interactive) - delay will hinder the experience

FTP (non-interactive) - delay will not be noticed, as files as being uploaded or downloaded

**Voice**

Smooth – uses a steady stream of bandwidth

Benign – opposite of greedy, they do not try to grab bandwidth from other applications

Drop-sensitive – if packets are dropped, the call quality degrades

Delay-sensitive – requires sub-150ms latency, otherwise the two ends of the calls are receiving the voice late

UDP – no ability to resend lost packets, best effort delivery

One-way requirements for Voice

Latency – <150ms

Jitter - <30ms

Loss <1%

Bandwidth (30-128kbps) (depends on codec)

**Video**

Bursty

Greedy

More movement requires more bandwidth

Delay-sensitive

Delay-sensitive

One-way requirements for Video

Latency – <150ms

Jitter - <30ms

Loss – 0.1 - 1.0%

Bandwidth (384kbps – 20+Mbps) (depends on resolution, color bitrate, fps, etc.)

Cisco Enterprise QoS Solution Reference Network Designs (SRND) Guide

This is a great resource

Loss - A relative measure of the number of packets that were not received compared to the total

number of packets transmitted. Loss is typically a function of availability. If the network is Highly

Available, then loss during periods of non-congestion would be essentially zero. During periods of

congestion, however, QoS mechanisms can determine which packets are more suitable to be

selectively dropped to alleviate the congestion.

Delay - The finite amount of time it takes a packet to reach the receiving endpoint after being

transmitted from the sending endpoint. In the case of voice, this is the amount of time it takes for a

sound to travel from the speaker’s mouth to a listener’s ear.

Delay variation (Jitter) - The difference in the end-to-end delay between packets. For example, if

one packet requires 100 ms to traverse the network from the source endpoint to the destination

endpoint and the following packet requires 125 ms to make the same trip, then the delay variation

is 25 ms.

Keys QoS Concepts for the CCNA

* Marking
* Device Trust
* Prioritization
  + Voice
  + Video
  + Data
* Shaping
* Policing
* Congestion Management

Cisco QoS Toolset

* Classification and Marking tools
* Policing and Markdown tools
* Scheduling tools
* Link-specific tools
* AutoQoS tools
* Call Admission Control tools

Chart

Description automatically generated

**Classification and Marking**

Label (a marking) applied to specific packets that tells each router and switch along the network path the priority level of the given packet.

Marking is done at Layer 2 or at Layer 3

When a marked packets arrive at a switch, the switch needs to decide what “class” the packets go into and the switch will then treat the packets per the class specifications.

Layer 2 Marking

EthII packet

|  |  |  |  |
| --- | --- | --- | --- |
| Dest Add | Source Add | 802.1Q | contains a 3bit priority header called COS – Class of Service | Voice Data |

COS (Class of Service) – has 0 to 7 as possible value.

000 – 0

001 – 1

110 – 5 Used by Voice

111 – 7

Phone, switch, and router need to agree of COS parameters

The problem with Layer 2, 802.1Q, specifying the COS is the link between the Switch and the Router needs to be a 802.1q Trunk connection. Therefore, you may want to use a Layer 3 marking.

Layer 3 Marking

Marking would be done with an IPv4/IPv6 Header

ToS (Type of Service) Field: (8 bits, first three are used)

Contains 8 binary values (0000 0000)

We used to only look at the most significant 3 bits (000x xxxx) – Values are 0 to 7 (voice is a 5)

The problem with this is there are only 8 options for ToS priority. The values were also not standardized

A value of 0 (000) would indicate best effort delivery

Often a value of 6 or 7 would be used for routing protocols

DSCP – 6 binary bits

Adjusted to use 6 binary bits (0000 00xx) and is known as DSCP  
**DSCP (Differentiated Service Code Points)** – same field in the IPv4 header, the interpretation of the bits is different, 6 bits are used instead of 3 bits (ToS) to determine how the packet is marked.

When 3 bits are used, it is known as IPP (IP Precedence)

When 6 bits are used, it is known as DSCP (Differentiated Service Code Points)

DSCP is backwards compatible to IP precedence.

The most significant 3 bits (IPP) are known as CS (Class Selector) Values

Voice data can be called IPP 5 (IP Precedence 5) or as CS 5 (Class Selector 5), it depends if they are using the old terminology or the new DSCP terms.

DSCP continued – (Note: in DSCP, Class Selector Values still exist) (Most significant bits are CS/IPP)

000 000 – when the 6 bits are set to 0, this denotes “best effort delivery”

001 000 – IPP 1 / CS 1

**Assured Forwarding Class**

* 001 001 – AF 1 1 Assured Forwarding Class 1 - 1
* 001 010 – AF 1 2 Assured Forwarding Class 1 - 2
* 001 110 – AF 1 3 Assured Forwarding Class 1 - 3
* 001 100 – AF 1 4 Assured Forwarding Class 1 - 4

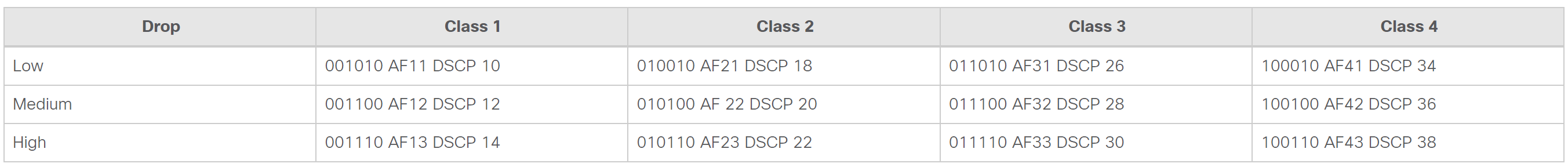
AF1 to AF4 (the digit refers to the binary value of the first 3 bits) (000 XXX) (0’s represents AF Values)

The first 3 binary digits (000 XXX) represent how important the traffic is (AF 1 is more important than AF 4)

The second digit (AF 1 **1**) represents how likely it is packets will be dropped

(Higher Number = More likely to be dropped)

**Assured Forwarding Values** (1st 3 bits = AF Class) (Bits 4+5 = Drop Probability) (Last bit = DS0 is always 0)



Diagram

Description automatically generated

Graphical user interface, text, application

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Within the ToS byte

The most significant 6 bits are used for DSCP (Differentiated Service Code Points) (aka DiffServ)

The least significant 2 buts are used for ECN (Explicit Congestion Notification) (ECN not in CCNA)

7 is the highest value for the most important network protocols (link layer and routing protocol keep alives)

**Expediated Forwarding (EF)**

RFC Definition (2598) – The EF PHB (per-hop behavior) can be used to build a low loss, low latency, low jitter, assured bandwidth, end-to-end service through DS (DiffServ) domains.

Such a service appears to the endpoints like a point-to-point connection or a ‘virtual leased line.’

This service has also been described as *Premium Service*.

DSCP value of 101110 (binary value) is recommended for the EF PHB, which corresponds to a DSCP value 46.

**IOS DSCP Settings**

Text

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Table

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Once you classified / matched on traffic, you need to do something with it; queue, prioritize, delay, drop, or rate-limit the classified/matched traffic.

Matching Criteria

CAR and class-based policing support different packet header values on which you can match to classify your traffic. Traffic matching defines the process of identifying traffic for rate limiting and/or packet marking.

Classification and Marking Tools

Classification and marking tools set this trust boundary by examining any of the following:

* Layer 2 parameters
  + 802.1Q Class of Service (CoS) bits,
  + Multiprotocol Label Switching Experimental Values (MPLS EXP)
* Layer 3 parameters
  + IP Precedence (IPP)
  + Differentiated Services Code Points (DSCP)
  + IP Explicit Congestion Notification (ECN)
  + Source/Destination IP address
* Layer 4 parameters
  + L4 protocol (TCP/UDP)
  + Source/Destination ports
* Layer 7 parameters
  + application signatures via Network Based Application Recognition (NBAR)

**Trust Boundaries**

The location in the network where packets are classified and marked.

Typically, a switch will not trust the ToS markings of a PC, but may trust the IP Phone with it ToS markings.

An IP phone will tell a switch, by using a marking, that its traffic is very important. The switch needs to be configured to trust that marking from the phone.

*Untrusted Domain:* The part of the network that you are not managing (e.g., Printer or PC)

From a QoS point of view, you are not going to trust the markings sent by a PC

*Trusted Domain:* The part of the network that only administrators can manage (e.g., Routers, Switches, Phones)

In an enterprise network, the trust boundary is usually at the edge of the network

For an ISP, the trust boundary is usually located at the last device they manage (router/switch)

By default, Cisco routers will override any QoS markings they receive on an untrusted boundary.

So, voice and data traffic will be treated the same if you don’t re-mark that traffic.

**Traffic Classes**

Before traffic policies can be applied to a packet, the packets need to be put into a class.

Classify and mark as close to the edge of the network as possible (best practice).

IP phones will mark their packets as they leave the IP phone.

For other traffic types you want to do your classification and marking on your edge switches.

Marking takes place on the edge, but every device along the path uses classification to determine what quality of service that traffic gets.

You can do your classification based on three criteria.

1. Marking – in a header (CoS or DSCP value)
2. IP Addressing – such as Destination/Source IP Subnet, Layer II Mac Address, Destination Port #, or domain address (cisco.com vs facebook.com)
3. Deep payload inspections using Application Signature - Network Based Application Recognition (NBAR)

**Network Based Application Recognition (NBAR)**

Uses Layers 4-7 and is more CPU intensive

Generally, only done at edge of network

2 Modes of Operation for NBAR

1. Passive Mode
   * Provides real-time statistics on application per protocol or interface and gives bidirectional statistics such as bit rate, packet, and byte counts.
2. Active Mode
   * Classifies application for traffic marking, so that QoS policies can be applied.

Policing and Shaping

Once traffic is identified it can be treated in a number of different ways, min bandwidth, max bandwidth, rate limit on bandwidth,

Policing and shaping limit the amount of traffic that you can transmit, aka they act as rate limiters.

Policers will drop excess traffic

Shapers will delay excess traffic

***Policers*** – perform check for traffic violations against a configured rate.

You can configure a policer to send the traffic without modification, re-mark the traffic and still transmit (provided you are below the bandwidth threshold), or drop the traffic the traffic.

There can be two traffic thresholds.

Tri-Color implementation

Below the threshold – transmitted as normal

Exceeds the first threshold – re-marked to a lower class and still transmitted

Exceeds the second threshold – dropped

***Shapers*** – Doesn’t drop the traffic, rather it smooths the traffic out by delaying the traffic. So, that after a period of time, the traffic falls within a specified bitrate.

Are usually used to meet Service Level Agreements. When the traffic spikes above the contracted rate, the excess traffic is buffered and is delayed until it falls below the contracted rate.

Policers are much harsher than Shapers, as they will drop traffic rather than delay traffic.

Shapers will attempt to smooth traffic out by buffering excess traffic.

Where to use Policers and Shapers

Policers are generally used as “Ingress Tools”. The traffic is dropped before it is processed (so you don’t waste resources). If a packet is going to be dropped anyway, it best to drop it on the ingress interface (incoming interface), so you don’t waste valuable bandwidth and CPU cycles.

Policers can be used on egress ports (exit interface) to control the traffic coming out of an interface,

Disadvantage of policers is that it is dropping packets which results in TCP resends. Doesn’t introduce jitter or delay, as they simply drop the packet.

Disadvantage of shapers in that they introduce jitter and delay when they slow down / buffer packets.

Shapers result in fewer TCP retransmissions.

**Queuing Mechanisms**

Congestion Management

Queuing or Buffering – determines the ordering of packets and the output buffers. Determines how traffic leaves a router or switch interface.

Round Robin Queuing Mechanism – All traffic is treated the same way (Real Time Traffic)

Strict Queuing Mechanism – High priority traffic is handled first. This can lead to starvation of bandwidth to lower priority traffic as all the bandwidth is going to higher priority traffic.

There are both Ingress and Egress Queuing Mechanisms

Egress Queuing Mechanisms

Queuing is only required when there is congestion. When queues fill up packets are reordered, so higher priority packets will leave the interface first.

* + Queuing – is the logic of ordering packets in output buffers
    1. Queuing is only activated when there is congestion on the link.
  + Scheduling – is the process of deciding which packet should be sent out next.
    1. Scheduling occurs regardless of whether there is congestion on the link .

FIFO (First in First out)

A single queue with packets that are sent in the exact order that they arrived.

Problem with this mechanism is that voice packets can be delayed by large data packets

Legacy mechanism

PQ (Priority Queue)

Consists of four queue’s (High, Medium, Normal, and Low) that are served in a strict priority order

The lower priority queues are served only when the high priority queues are empty.

Problem with this mechanism is it can lead to starvation in lower priority queues.

CQ (Custom Queuing)

Consists of 16 queues serviced in a round-robin fashion.

In order to prevent starvation, it provides traffic guarantees.

Problem with this mechanism is that it doesn’t provide priority to real-time traffic and introduces delay

If you have important voice traffic arriving it will only be serviced in its round

WFQ (Weighted Fair Queuing)

Algorithm that divides the internet bandwidth by the number of flows.

Provides a good service for real-time traffic.

No bandwidth guarantees for particular flows.

Some flows can starve other flows.

WFQ Scheduling Algorithm

Incoming packets are classified by flows rather than classes.

Flows are classified by source/destination IP Address, the protocol, and a port number.

A “weight” is added to a “flow” based on certain criteria (IPP or RSVP - Resource Reservation Protocol)

Prioritizes smaller packets over larger packets.

More Fair queuing algorithm, in that it provides better QoS for small packets (which are generally used for interactive sessions) (example: voice packet may be 20 bytes in size, while an FTP packets may be 1500 bytes)

You can increase prioritization by adding a weight to smaller packets (based on IPP for example)

Uses a clever Scheduling Algorithm to prioritize smaller interactive packets, which you can make appear even smaller by increasing the IP Precedence (IPP) of the packet.

Problem with this mechanism is that it doesn’t provide Bandwidth Guarantees.

CBWFQ (Class-based Weighted Fair Queuing)

* Guarantees bandwidth to specific classes and provides dynamic fairness of other flows.
* Allows you to create different classes where you can specify a min bandwidth
* Weighted Fair Queuing can be used on the “Best Effort Class” to ensure that traffic is handled fairly.
* Traffic gets fair bandwidth guarantees
* Minimum bandwidth to HTTP, FTP, Voice traffic, Video traffic, etc.
* No Latency guarantees (aka there is no priority queue. Thus, it suitable only for data networks.

LLQ (Low Latency Queuing)

* Basically, is Class-based WFQ with an added priority queue for real-time traffic.
* The priority queue has a minimum bandwidth guarantee, but is also policed (has a max bandwidth limit)
* Minimum bandwidth guarantee for voice and guarantees voice traffic won’t starve by having a max bandwidth rate limit for other traffic types.

Ways to Avoid Congestion

Queues on routers and switches are finite, they can only hold or buffer a certain number of packets.

If there is a burst of traffic and the buffers are overrun (more packets are enqueued than dequeued), it will start to drop packets.

Tail Drop – when the queue fills up, all new packets will be dropped.

WRED (Weighted Random Early Detection)

Starts randomly dropping packets from multiple flows before the queue fills up to avoid congestion.

Better utilization of an interface’s bandwidth, as some TCP flows are slowing down while others are speeding up simultaneously. Which in aggregate gives you a better utilization of the interface.

Below the threshold – transmitted as normal, no packets are dropped

Exceeds the minimum threshold – random drops of packets

Exceeds the maximum threshold – Full drops of a traffic class

The Weighted part of WRED has to do with our ability to apply a weight to certain types of traffic. E.g., drop FTP traffic before dropping HTTP traffic. Preselect which packets will get dropped.

Typically, you only want to drop TCP packets because TCP flows will retransmit.

Allows for buffer space to be left for voice packets.