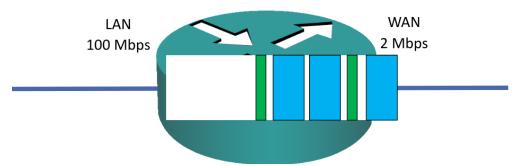
Quality Requirements for Voice and Video

- Voice and traditional standard definition video packets must meet these recommended requirements to be an acceptable quality call:
 - Latency (delay) ≤ 150 ms
 - Jitter (variation in delay) ≤ 30 ms
 - Loss ≤ 1%
- These are one way requirements, meaning a packet sent from a phone in HQ has 150ms to reach the phone in the branch, and vice versa
- HD video has stricter requirements

Congestion

- Packets are arriving faster than they can be sent out
- Packets wait in the queue to go out
- Packets are sent out FIFO in the order they were received



- Congestion causes delay to packets as they wait in the queue
- As the size of the queue changes it causes jitter
- There is a limit to the size of the queue. If a packet arrives when the queue is full the router will drop it
- Voice and video calls (and applications) will be unacceptable quality if they do not meet their delay, jitter and loss requirements

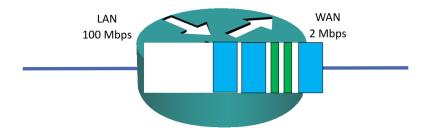
Congestion Example with QoS Queuing

Traffic destined for the branch comes in on the LAN interface at a rate higher than 2 Mbps



Congestion Example

- Packets are arriving faster than they can be sent out
- Packets wait in the queue to go out
- The router recognises the voice packets and moves them to the front of the queue to minimise their delay



Effects of QoS Queuing

- QoS queuing can reduce latency, jitter and loss for particular traffic
- The original driver for QoS was Voice over IP but it can also be used to give better service to data applications
- If you're giving one type of traffic better service on the same link you started with, the other traffic types must get worse service
- The point is to give each type of traffic the service it requires
- QoS queuing is not a magic bullet and is designed to mitigate temporary periods of congestion. If a link is permanently congested the bandwidth should be increased

Classification and Marking

- For a router or switch to give a particular level of service to a type of traffic, it
 has to recognise that traffic first
- Common ways to recognise the traffic are by COS (Class of Service) marking, DSCP (Differentiated Service Code Point) marking, an Access Control List, or NBAR (Network Based Application Recognition)

Layer 2 Marking - CoS Class of Service

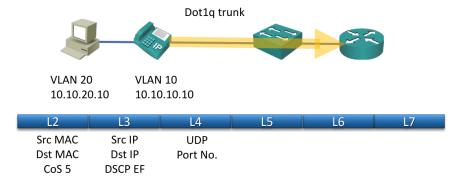
- There is a 3 bit field in the Layer 2 802.1q frame header which is used to carry the CoS QoS marking
- A value of 0 7 can be set. The default value is 0 which is designated as Best Effort traffic
- CoS 6 and 7 are reserved for network use
- IP phones mark their call signalling traffic as CoS 3 and their voice payload as CoS 5

Layer 3 Marking - DSCP

- The ToS Type of Service byte in the Layer 3 IP header is used to carry the DSCP QoS marking
- 6 bits are used which gives 64 possible values. The default value is 0 which is designated as Best Effort traffic
- IP phones mark their call signalling traffic as 24 (CS3) and their voice payload as 46 (EF)
- There are standard markings for other traffic types, such as 26 (AF31) for mission critical data, and 34 (AF41) for SD video

The Trust Boundary

The switch should be configured to trust markings from the IP phone and pass them on unchanged, but mark traffic from the PC down to CoS 0 and DSCP 0



Recognising Traffic with an ACL

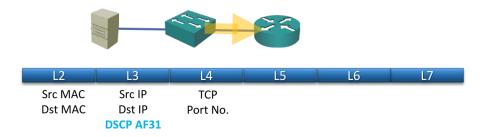
- An Access Control List can be used to recognise traffic based on its Layer 3 and Layer 4 information
- For example SSH traffic going to and from the router 10.10.100.10 on TCP port number 22

Recognising Traffic with NBAR

- NBAR (Network Based Application Recognition) can be used to recognise traffic based on its Layer 3 to Layer 7 information
- Signatures can be downloaded from Cisco and loaded on your router which recognise well known applications

Classification and marking

- DSCP is the preferred classification and marking method because the router can very quickly gather the information from a single byte in the IP header
- If using another method such as an ACL or NBAR is being used, this should be done as close to the source as possible and then a DSCP value added



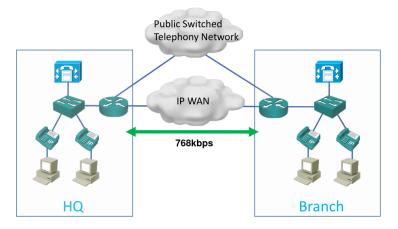
Congestion Management

- Queuing can be used to manage congestion on routers and switches
- CBWFQ (Class Based Weighted Fair Queuing) gives bandwidth guarantees to specified traffic types
- LLQ (Low Latency Queuing) is CBWFQ with a priority queue
- Traffic in the priority queue is sent before other traffic

MQC Modular QoS CLI

- Cisco QoS configuration uses the MQC Modular QoS CLI
- It has 3 main sections
- Class Maps define the traffic to take an action on
- Policy Maps take the action on that traffic
- Service Policies apply the policy to an interface

Congestion Management Example



- 768kbps WAN Link between offices
- Need to support 10 concurrent voice calls over the WAN
- Each call = 25.6kbps
- · 256kbps provisioned for voice calls
- · 512kbps provisioned for data
- Data will sometimes burst above 512kbps creating congestion

(This config isn't required on the CCNA)

Congestion Management Example - LLQ

- Configure the same LLQ policy on the routers in HQ and the branch
- Apply to the WAN interfaces

```
class-map VOICE-PAYLOAD
match ip dscp ef
class-map CALL-SIGNALING
match ip dscp cs3
!

policy-map WAN-EDGE
class VOICE-PAYLOAD
priority percent 33
class CALL-SIGNALING
bandwidth percent 5
class class-default
fair-queue
!
interface Serial0/0/0
bandwidth 768
service-policy out WAN-EDGE
```

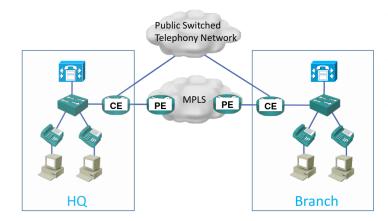


Shaping and Policing

- Traffic Shaping and Policing can be used to control traffic rate.
- They both measure the rate of traffic through an interface and take an action if the rate is above a configured limit.

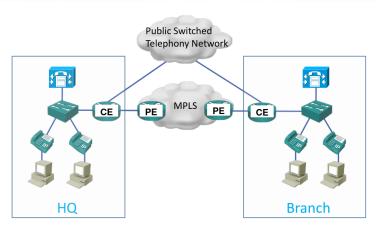
- Traffic shaping buffers any excess traffic so the overall traffic stays within the desired rate limit.
- Traffic policing drops or re-marks excess traffic to enforce the specified rate limit.
- Classification can be used to allow different rates for different traffic types.

Policing Scenario – Service Provider PE



- In this example the customer has provisioned an MPLS VPN with a service provider
- The physical links from the CE to PE routers are 100 Mbps, but the customer has paid for 10 Mbps in their SLA
- The service provider enables policing inbound on the PE routers to limit the customer to 10 Mbps bandwidth.
 Excess traffic is dropped

Shaping Scenario – Customer CE



- The CE to PE link is 100 Mbps. If the customer sends at a rate above 50 Mbps, excess traffic will be dropped by the provider
- Some traffic would reach the destination, some would not
- The dropped traffic would be random, it could be data or voice
- When voice packets are dropped call quality is unacceptable
- The customer configures shaping outbound on the CE WAN interfaces, with nested LLQ

Shaping Example

- 10 Mbps SLA on WAN outside interface
- 100Mbps LAN inside interface
- 1 Mbps provisioned for voice
- 3 Mbps provisioned for video
- 6 Mbps provisioned for data
- Data will sometimes burst above 6 Mbps creating congestion

class-map VOICE
match ip dscp ef
class-map VIDEO
match ip dscp af41
class-map SIGNALLING
match ip dscp cs3

policy-map NESTED
class VOICE
priority 1024
class VIDEO
priority 3072
class SIGNALING
bandwidth 128
class class-default
fair-queue

policy-map WAN-EDGE
 class class-default
 shape average 10000000
 service-policy NESTED

Interface FastEthernet0/0
service-policy out WAN-EDGE