UNIT - V

Multimedia Networking:-

Multimedia networking:

* People in all concers of the world are currenty useing the Internet to watch movies and television shows on demand

* Internet movie & television distribution companies such as Netfix and Hulu in North America and Youku & kankan in China have partically become household names.

Network Application.

* Skype, google talk, and QQ (enormously popular in China) allow people to not only make "telephone calls" over the internet but to also enhance those calls with video and multiperson conferencing.

* Wireless devices connected to the internet via 4G & WIFI access N/w.

Multimedia N/w applications:

* It is any N/w application that employs audio & video.

* It is useful to consider the intrinsic characteristics of the audio and video media themselves.

a) properties of video . high bit rate.

b) properties of Audio - Lower bandwidth requirement than video

types:-

1) Streaming stored Audio & video

key words:-
Streaming
Interactivity,
Continuous playout

2) Conversational voice and video over IP

It is referred to as Internet telephony.

3. Streaming live audio & video

It is similar to traditional Broadcast radio & television. except that transmission takes place over the internet.

Streaming Stored video.

* Prerecorded videos are placed on servers and users send request to these servers to view the video on demand.

* It can be classified into three categories such as

* UDP Streaming.
* HTTP Streaming.
* Adaptive HTTP Streaming.

UDP Streaming:-

* The Server transmits video at a rate that matches the client's video consumption rate by clocking out the video chunks over UDP at a steady rate.

HTTP streaming:

* The video is simply stored in an HTTP server as an ordinary file with a specific URL.

* When a user wants to see the video, the client establishes a TCP connection with the server and issues an "HTTP GET" request for the URL.

* The server then sends the video file, within a HTTP response message as quickly as possible.

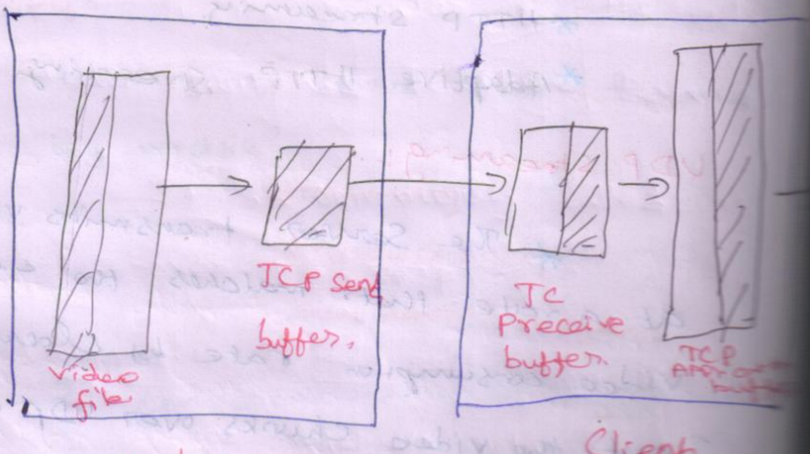


Fig. Streaming stored video over HTTP/TCP

* Prefetching video
* Client application buffer & TCP buffers.
* Analysis of video Streaming
* Early termination & Repositioning the video

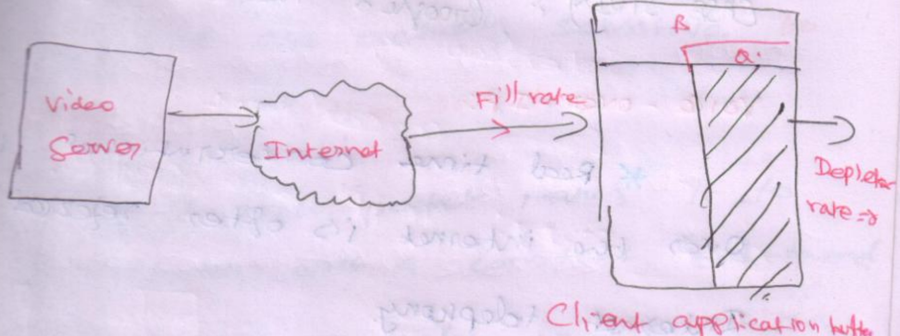


Fig: Analysis of Client-side buffering for video streaming.

$$\boxed{\text{Buffering delay } (t_p) = Q/x}$$

Adaptive Streaming and DASH.

* Dynamic Adaptive Streaming over HTTP (DASH).
* The video is encoded into several different versions, with

each version having a different bitrate and corresponding a different quality level.

* It allows clients with different Internet access rates to stream in video at different encoding rates.

CASE STUDY : Google's Network Infrastructure

Voice - over - IP :

* Real time conversational voice over the internet is often referred to as Internet telephony.

* It is similar to the traditional circuit - switched telephone service.

* It is commonly called as Voice-over-IP (VOIP).

limitation of the best effort IP service

* The internets network layer protocol, IP provides best effort services.

* The service makes its best effort to move each data stream from source to destination as quickly as possible.

* The lack of such guarantees posses significant challenges to the design of real time conversational Application.

* While are actually sensitive to packet delay, jitter and loss.

* If each packet makes it to the receiver with a constant end-to-end delay, then packets arrive at the receiver periodically every 20ms.

* Packet loss
* end to end delay
* packet jitter.

Removing jitter at the Receiver for Audio

* The following two mechanisms

1. prepending each chunk with a timestamp

2) Delaying playout of chunk at the receiver.

Fixed playout Delay

Adaptive playout delay

Removing from packet Loss:

* Forward Error correction (FEC)

* Interleaving.

* Error concealment

CASE STUDY: VOIP with Skype.

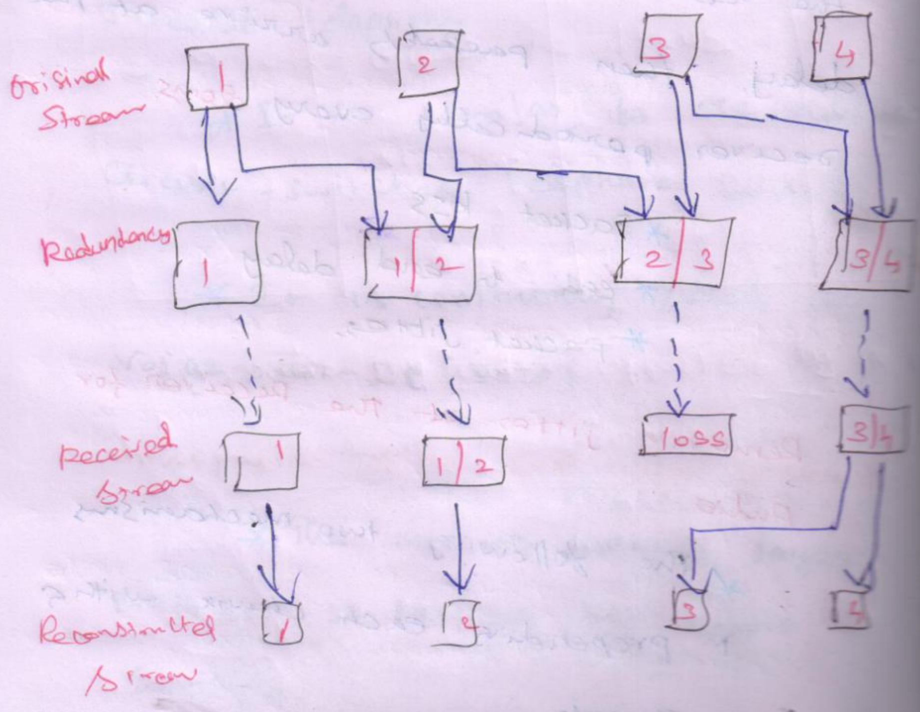


Fig. Piggy backing lower quality redundancy including VoIP and "video conferencing" are compelling and very popular.

protocols types:-

1. Real time protocol (RTP)

2. Session Initiation protocol (SIP)

RTP: Basics.

* RTP header
* RTP packet
* RTP session.

RTP packet Header fields:-

→ Sequence number field.

* It is 16-bit long.

→ Time stamp field.

* It is 32 bits longs.

→ Synchronization Source identifier

* It is 32 bits long.

Pay load Type	Sequence number	Time Stamp	Synchronizer Source identifier	Miscellaneous fields

Fig. RTP header fields.

Payload Type number	Audio format	Sampling rate	Rate
0	PCM M-low	8KHZ	64kbps
1	1016	8KHZ	4.8kbps
3	GSM	8KHZ	13 kbps
7	LPC	8KHZ	64kbps
9	G.722	16KHZ	.68.64kbps
14	MPEG Audio	90KHZ	–
15	G.728	8KHZ	16k[illegible]

Fig. Audio pay load Supported by RTP

Payload type number.	video format
26	Motion JPEG.
31	H.261
32	MPEG 1 video
33	MPEG 2 video

Fig. Some video payload types supported by RTP.

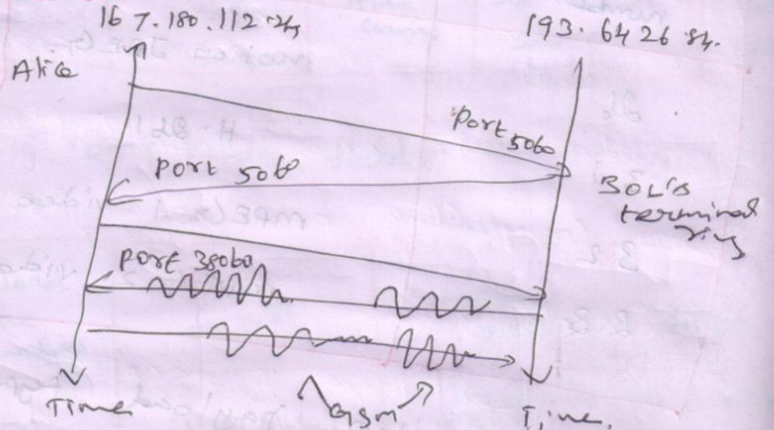
SIP:-

* It is defined in RFC 3261 RFC 5411 is an open and light weight protocol that does the following.

1) It provides mechanisms for establishing calls b/w a caller and a callee over an IP Network.

2) It provides mechanisms for the caller to determine the current IP address of the callee.

Setting up a call to a known IP address.



* It provides mechanism for call management, such as "adding new media streams during the call, changing the encoding during the call, inviting new participants during the call, call transfer and call holding".

* SIP address.

* SIP message.

* Name translation & user location

* media is sent directly between the two clients.

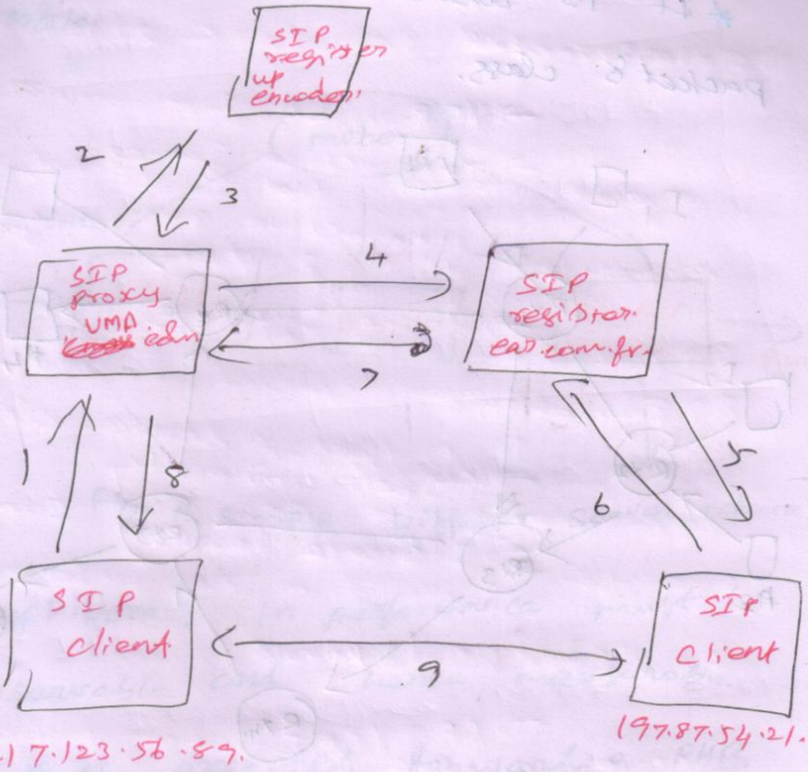


Fig. Session Initiation, Involvig SIP proxies and registrars.

This media is also an SIP acknowledgement message.

* When a DS-marked packet arrives at a differentiated service capable router, the packet is forwarded onto its next hop according to the so called as per-hop-behaviour (PHB).

with in the ~~ng~~ negotiated traffic profile.

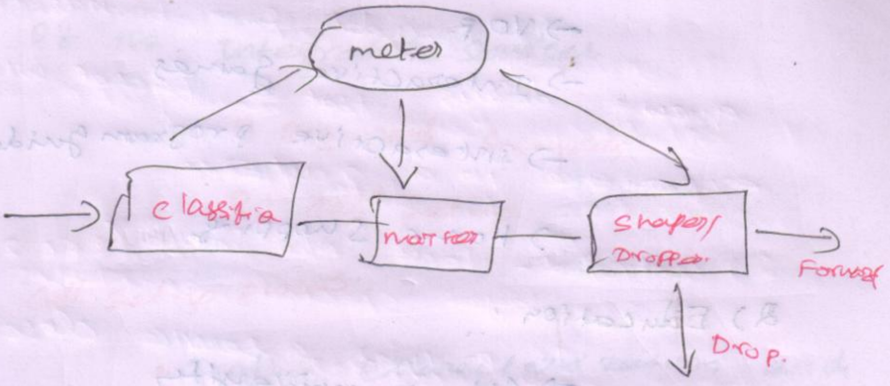


Fig. A simple Different Server network.

* Differences in performance must be observable and hence measurable.

* The expected forwarding PHB speacities that 'the departure rate of a class of traffic from a router must equal (or) exceed a configured rate.

* The assured forwarding PHB divide traffic into four classes, where each AF class is guaranteed to be provided with some minimum amount of bandwidth & buffering.

individual application session

* Two key features lie at the Leare of the Integrated Services.

1) Reserved Resources.

2) call Setup.

Call set up process:-

* Traffic Characterization and Specification of the desired QoS.

~~eg:- tspec~~:

* TSPEC:- Characterizes the traffic the sender will be sending into the networ (RFC 2215)

* RSPEC:- Characterizes the QoS being requested by the Connection (RFC 2210)

* signaling for call setup.

* Pre-element calls admission.

Service classes:-

* The integrated Services ~~arch~~ architecture defines two major

service classes. They are

1) Guaranted QoS. (RFC 2212)

It provides firm bounds on queuing delays that a packet will experience in a router.

2) controlled - load N/w Service. (RFC 2211)

It provides a service closely approximating the QoS that same flow would receive from an unloaded Network element.

RSVP (Resource Reservation protocol).

* It allows applications to reserve bandwidth for their dataflows

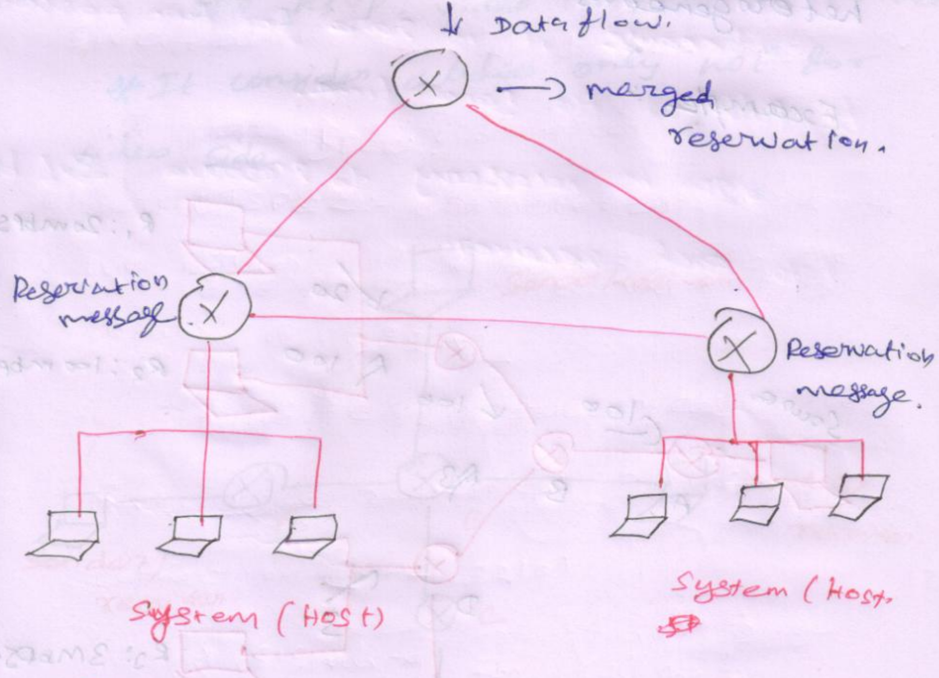
* To implement RSVP software must be present on the receivers, senders and routers.

* These are differences in performance must be observable and hence measurable.

Principle characteristics:-

* It provides reservation for bandwidth in multicast trees.

* It is receiver - oriented, that is the receiver of the data flow initiates and maintains the resources reservation used for that flow.



Session:-

* It is consist of multiple data flows.

* The specific data flow is identifier by a flow identifier field

* RSVP is sometimes referred to as a signaling protocol.

* It meanings that is allows hosts to establish a tear down approach reservation for data flows

* RSVP layers special emphases on heterogeneous receivers.

Example:

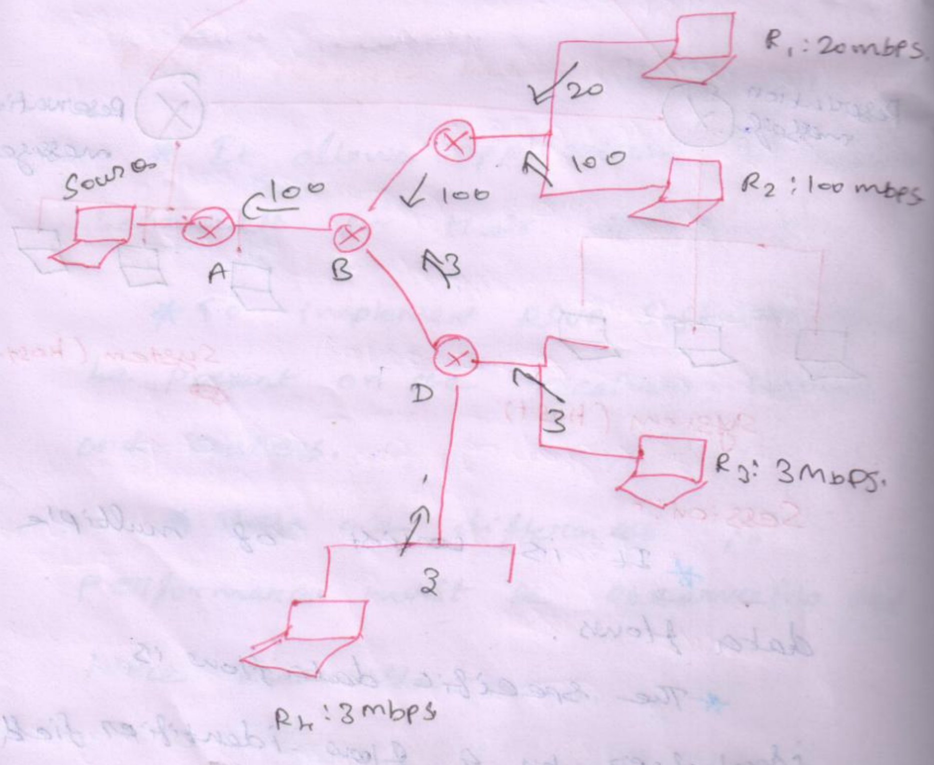


Fig. Sample scenario for RSVP.

* Source is transmitting a Sporting event (video and Audio).

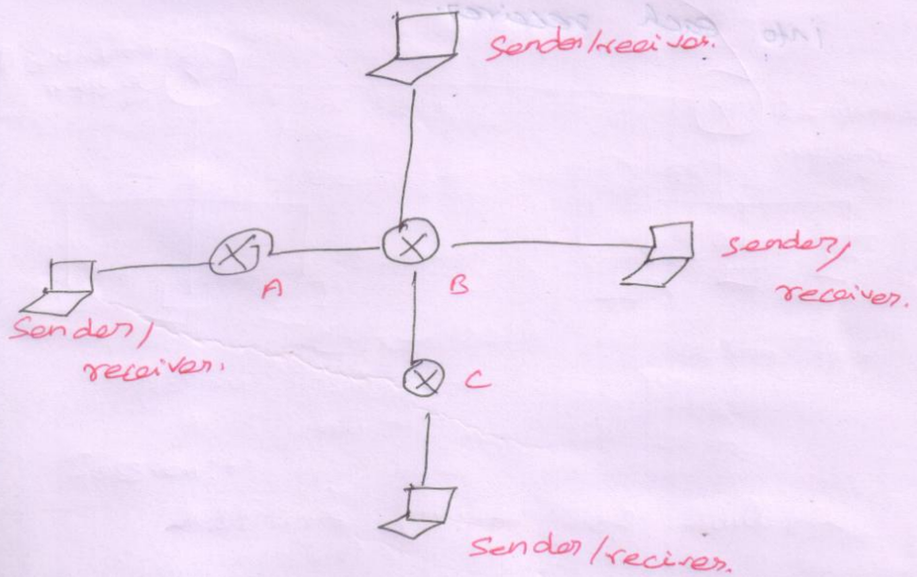
* Use layered - coding to handle hetero geneous receivers.

* Receiver - oriented reservation protocol.

Example 2 :

For RSVP : video conferencing. protocol.

* It consider audio only not for video side.



* Each person has 3 windows open to look at 3 others.

* Each person wants to see each of the video at 3 mbps.

* RSVP would reserve 9 mbps in one direction and 3mbps in the other.

Consider audio only

* Suppose 'b' bits per second are needed for each stream.

* Because it is rare for two persons to talk at the same time.

* It is necessary to reserve 2xb bps into each receiver.