

Multimedia over the Internet

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Who am I?

François Fluckiger

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Geneva, Switzerland

Birth place of the World-Wide Web

Head, Internet Infrastructure

Part-time Professor of Computer Science at the

University of Geneva

For further reading ...

- Reference book:

“Understanding Networked Multimedia”

Prentice Hall ISBN 0-13-190992-4

by F.Fluckiger

Where to find more in the Book?

See pointer at the bottom

Part 1

Requirements of Multimedia over the Internet

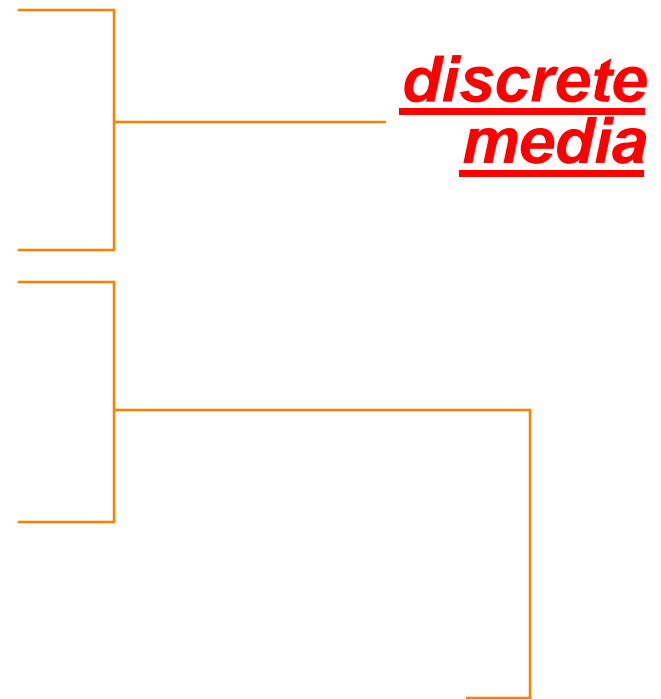


- **Key requirements**
- **Bit rate requirements**
 - Audio requirements
 - Video requirements
- **Delay requirements**
 - Jitter
 - Inter-media synchronization
- **On compression ...**

Multimedia characteristics

■ Information types

- Text
- Images (pixel matrix)
- Graphics (logical objects)
- **Animation** (moving graphics)
- **Video** (moving images)
- **Audio**



- **continuous media** (time is part of their semantics)

Audio/video network requirements

- **Key requirements**

- Bit rates
- Transit delay **variation**
- Multicasting capabilities (for distribution)

- **Other requirements**

- Transit delay
- Error rate

- **Key requirements**



- **Bit rate requirements**

- **Audio requirements**
- **Video requirements**

- **Delay requirements**

- **Jitter**
- **Inter-media synchronization**

- **On compression ...**

Types of applications

- **Traditional real-time applications** e.g. PABXs

constant bit rate (CBR)

- **Traditional bulk data applications** e.g. file transfer, email

available bit rate (ABR)

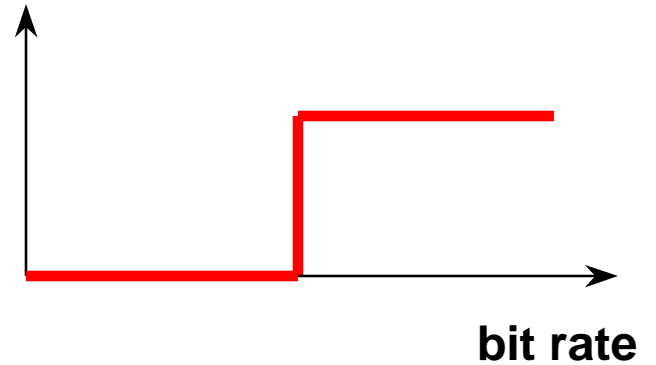
- **Modern real-time applications** e.g. compressed audio, video

variable bit rate (VBR)

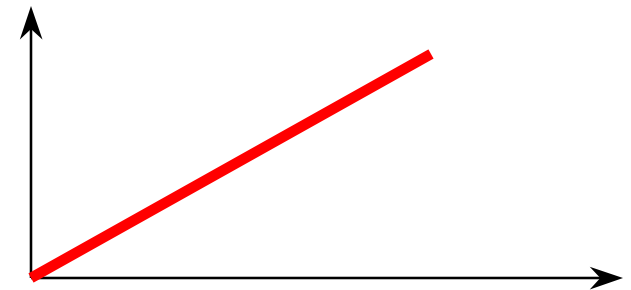
Quality of Service and bit rate

- **CBR** applications

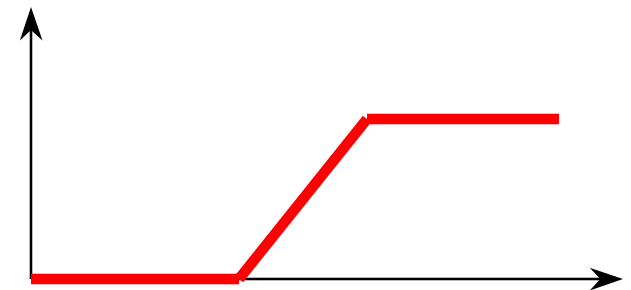
User satisfaction



- **ABR** applications



- **VBR** applications



From S. Shenker,
Fundamental Design Issues for the Future Internet, 1995

Principle (or platitude)

**The grass is always greener on
the other side of the hill ...**

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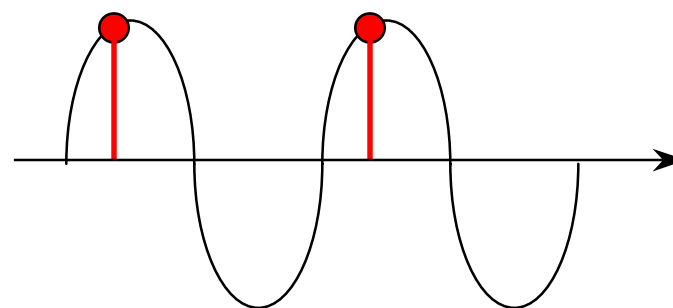
Nyquist theorem

- To faithfully represent an analog signal

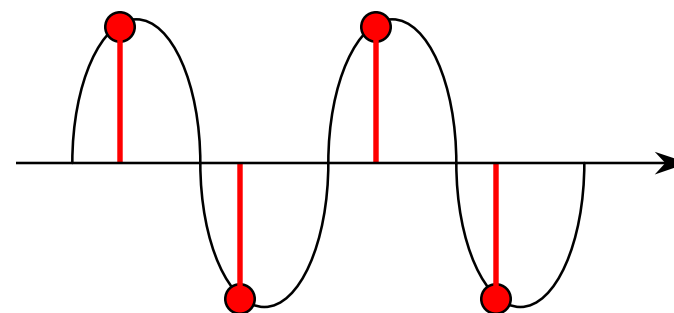
- if maximum frequency f
- sampling rate at least $2f$

- Application to audio

- if sampling rate is **8 kHz**
- bandwidth is **3.4 kHz**



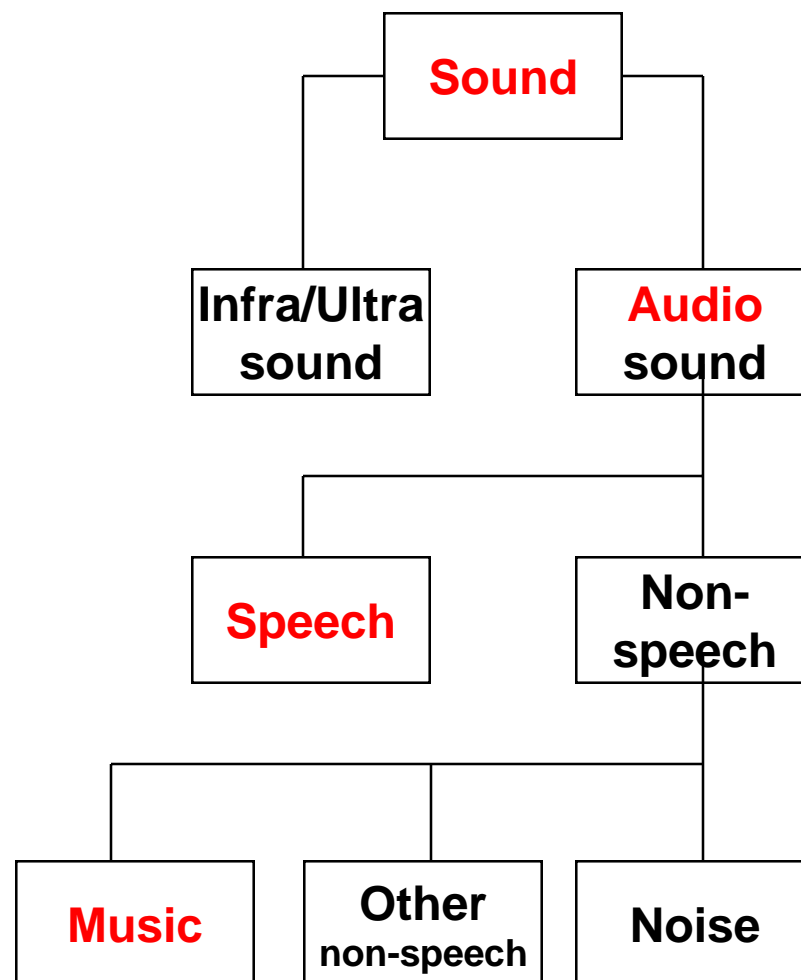
Sampling at f :
impossible to reconstruct



Sampling at $2f$:
easier to reconstruct

Sound, Audio, Speech, ...

- **Sound**: vibration of matter
- **Audio**: audible sound (by humans)
 - human audible spectrum:
20 Hz - 20kHz
- **Speech**: a particular type of sound
 - we hear better than we talk
 - speech spectrum:
50-10 kHz
- **Music**: a particular case of non-speech sound



Audio bit rate requirements

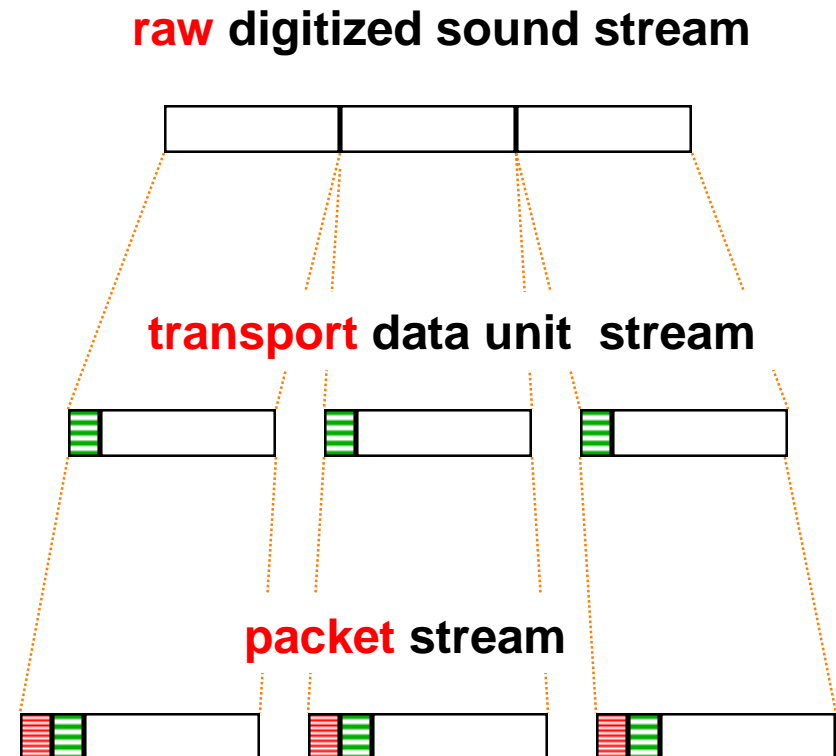
Quality	Technique or standard	Kbps	Compr.
■ Telephone quality			
■ Standard	G.711 PCM	64	
■ Standard	G.721 ADPCM	32	Y
■ Lower	G.728 LD-CELP	16	Y
■ Lower	GSM	13	Y
■ Standard-	G.729 LD-CELP	8	Y
■ Lower+	CELP	5-7	Y
■ CD Quality			
■ Consumer CD-audio	CD-DA	1441 (stereo)	
■ Consumer CD-audio	MPEG with FFT	192-256	Y
■ Sound studio quality	MPEG with FFT	384	Y
■ Consumer CD-audio (MP3)	MPEG2.5 Layer III	128 (stereo)	Y

Which bit rate is actually needed?

■ Network overheads incl.:

- RTP header (12 bytes)
- Transport Protocol header (usually UDP, 8 bytes)
- IP header (20 bytes)
- Example:
raw G.711 **64** Kbps requires from **68** to 80 Kbps

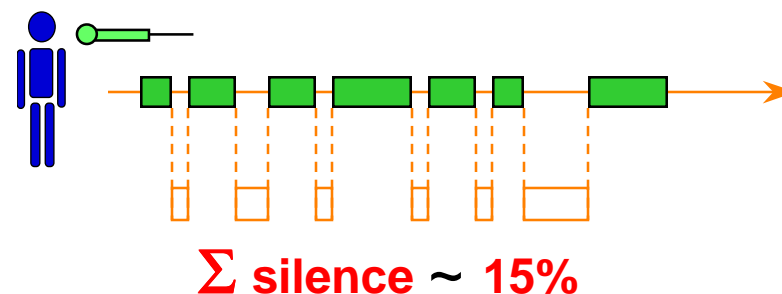
■ *However, speech contains silence*



Silences in speech

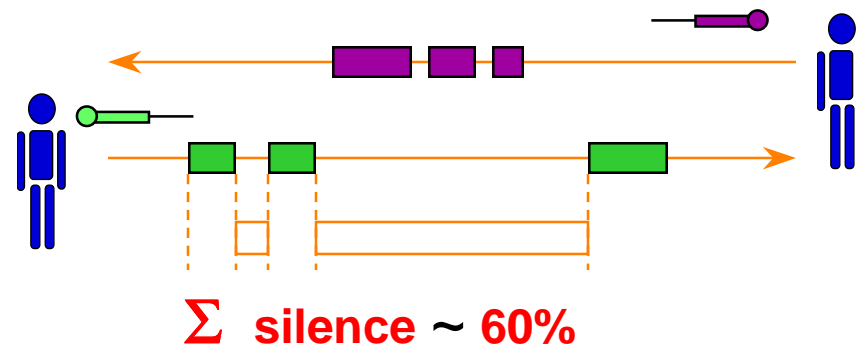
■ Monologue

- typically 15% silence



■ Bi-party telephone conversation

- 20% silence for overall conversation
- 60% silence for each party
- If silence suppressed, required bit rate is in effect <40% of nominal raw bit rate



Observations, Trends

Audio does not eat
bandwidth

Voice packets will swim in an
ocean of data packets

- **Key requirements**
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Video bit rate requirements

Quality	Technique or standard	Mbps	Compr.
■ Video conf. quality	H.261	0.1	Y
■ VCR quality	MPEG-1	1.2.	Y
■ Broadcast quality	MPEG-2	2-4 (1)	Y
■ Studio-quality digital TV			
■ Uncompressed	ITU-R 601	166	
■ Compressed	MPEG-2	3 to 6 (2)	Y
■ HDTV			
■ Uncompressed	CD-DA	2000	
■ Compressed	MPEG-2	25 to 34	Y

(1): future; current implementations: 4 to 7

(2): future; current implementations: 6 to 10

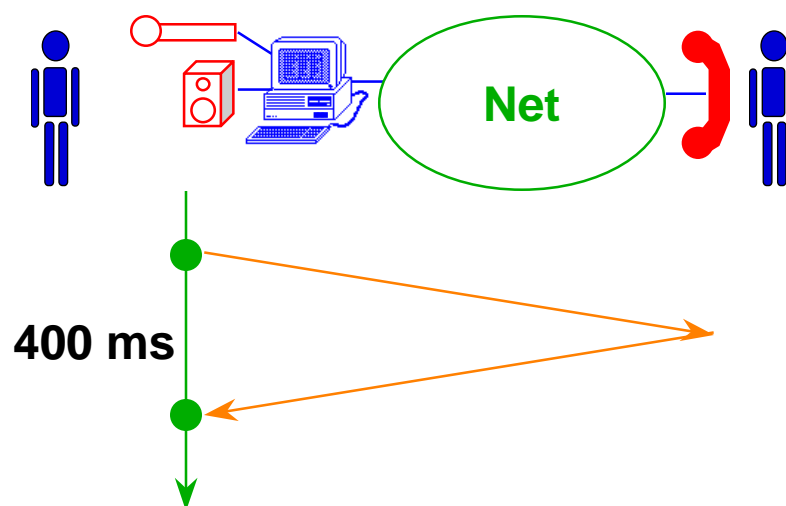
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Network Transit Delay

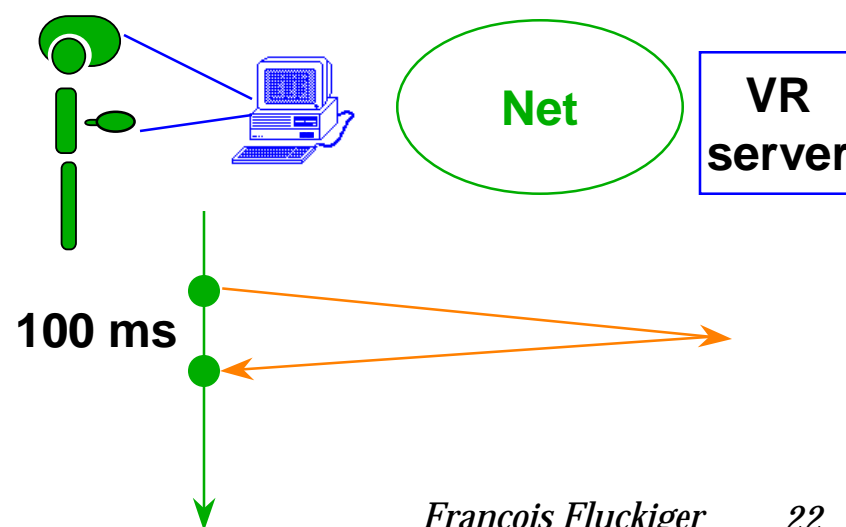
■ Telephone conversation:

- Round-trip delay < 400 ms
- for natural conversation



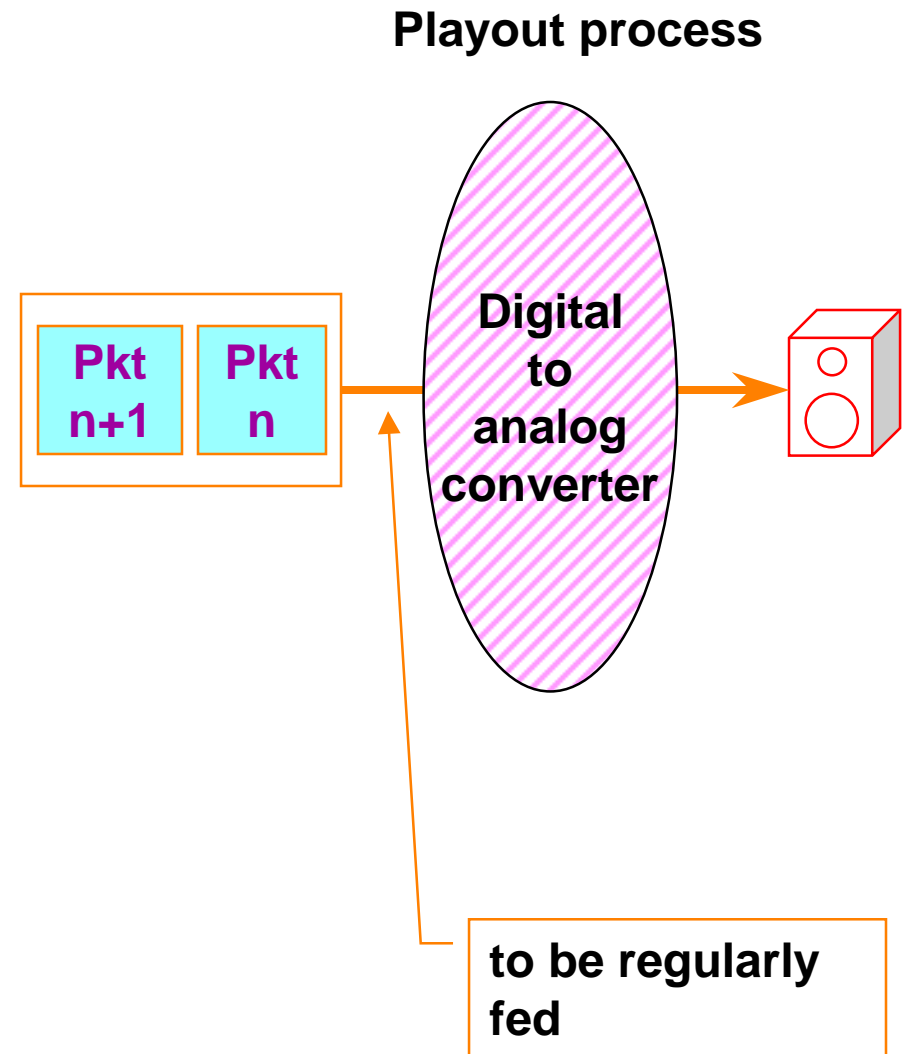
■ Virtual reality

- Round-trip delay < 100 ms
- for impression of immersion



Transit delay variation (Jitter)

- Receiver to wait a **delay offset** before playout
- Called **delay equalization**
- Increases overall end-to-end latency

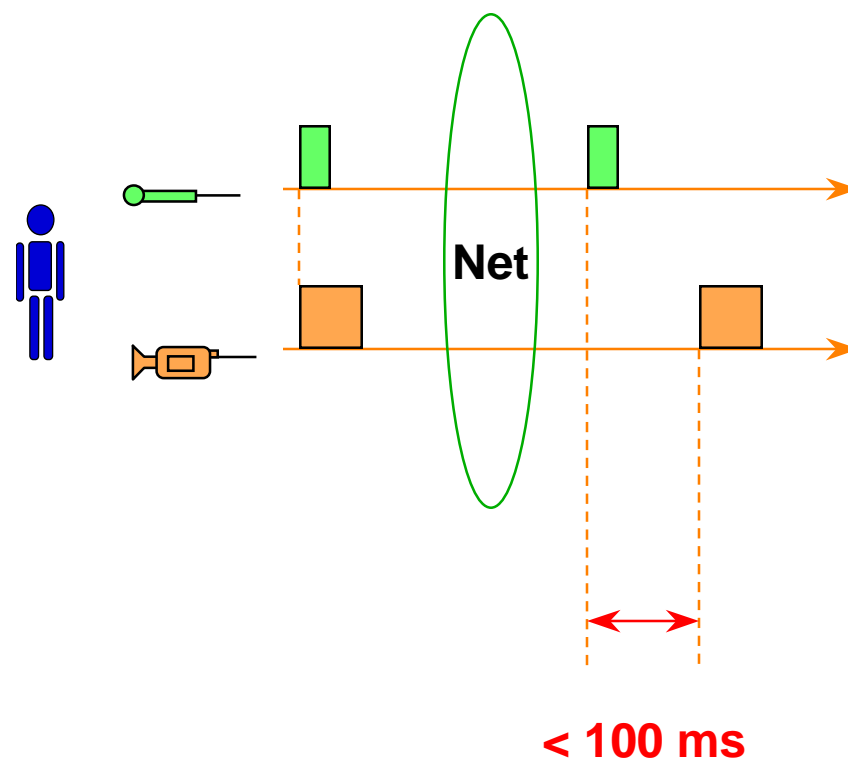


- **Key requirements**
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- **On compression ...**



Inter-media synchronization

- Called orchestration
- Particular case:
lip synchronization
- A skew of **80-100 ms** is generally tolerated



Audio/video relative priorities

- The ear behaves as a *differentiator*
- The eye behaves as an *integrator*
- Toleration of transmission errors affecting sound much lower than for video
- When audio and video streams part of the same application competing for network resources, **audio stream should have priority**

- **Key requirements**
- **Bit rate requirements**
 - Audio requirements
 - Video requirements
- **Delay requirements**
 - Jitter
 - Inter-media synchronization



- **On compression ...**

Audio-compression techniques

Encoding techniques

- DPCM, Delta
- ADPCM

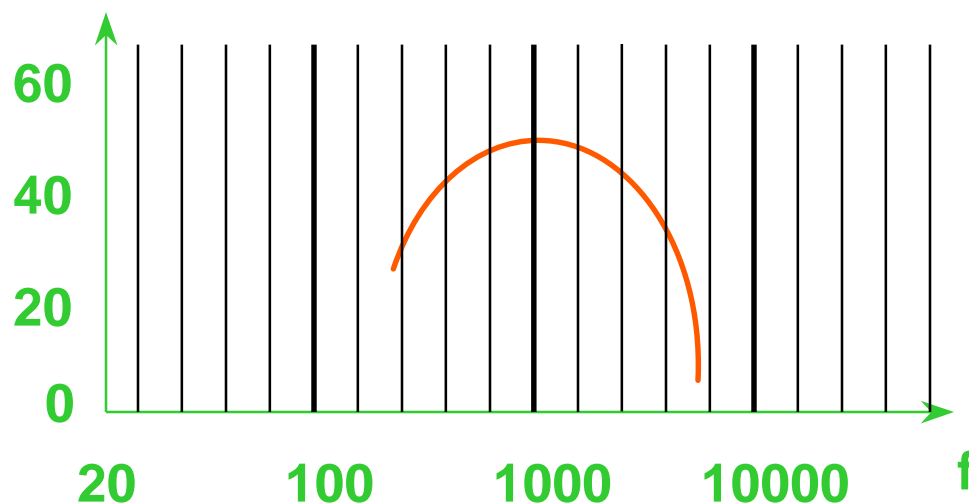
Source compression techniques

Based on psycho-acoustic model

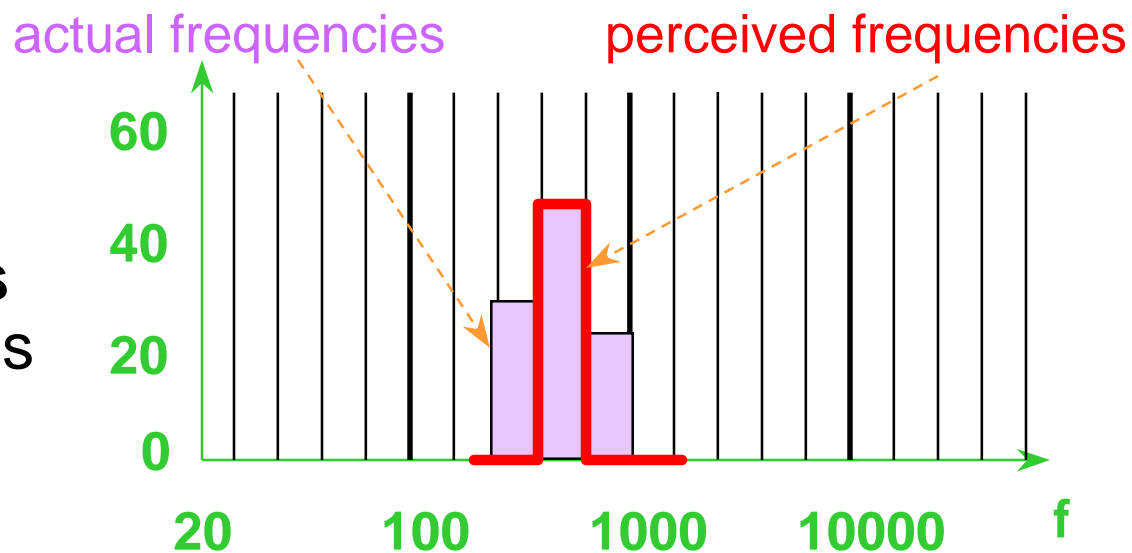
- **Transform encoding** (all sounds)
 - Fast Fourier Transform (FFT)
 - Wavelet transform
- **Source modeling/synthesis coding** (for speech)

Psycho-acoustic and Masking

- **Response of ear to frequency:**
 - ear most sensitive between 2 - 5 kHz

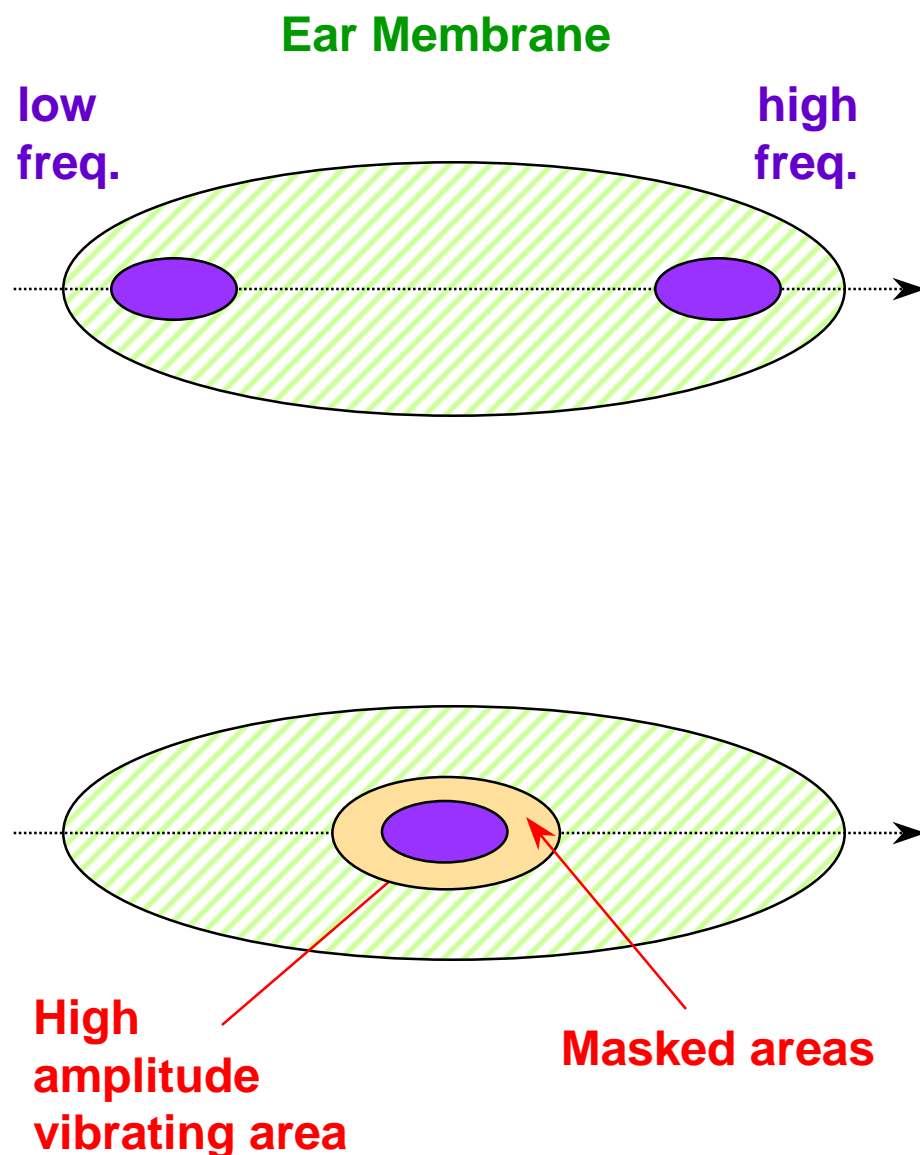


- **Masking:**
 - ear does not register energy in **some frequencies band**, when there is more energy in a nearby band



Physiology and masking

- Ear membrane vibrates as a function of frequency
 - **High** frequencies:
 - at one end
 - **Low** frequencies:
 - at opposite end
- Vibration of a area **forces close areas to vibrate at the same frequency**, and not at their own



Voice modeling techniques

- **Human Vocal system model relies**
 - on a **set of cylinders** of differing diameters
(e.g. 10 in LPC-10)
 - excited by a signal at a certain frequency

- **Operates over 20 ms, on standard PCM samples**

Principle (or platitude)

This is **what we perceive** that count,
not what the physical reality is!

or

The Reality is what we perceive

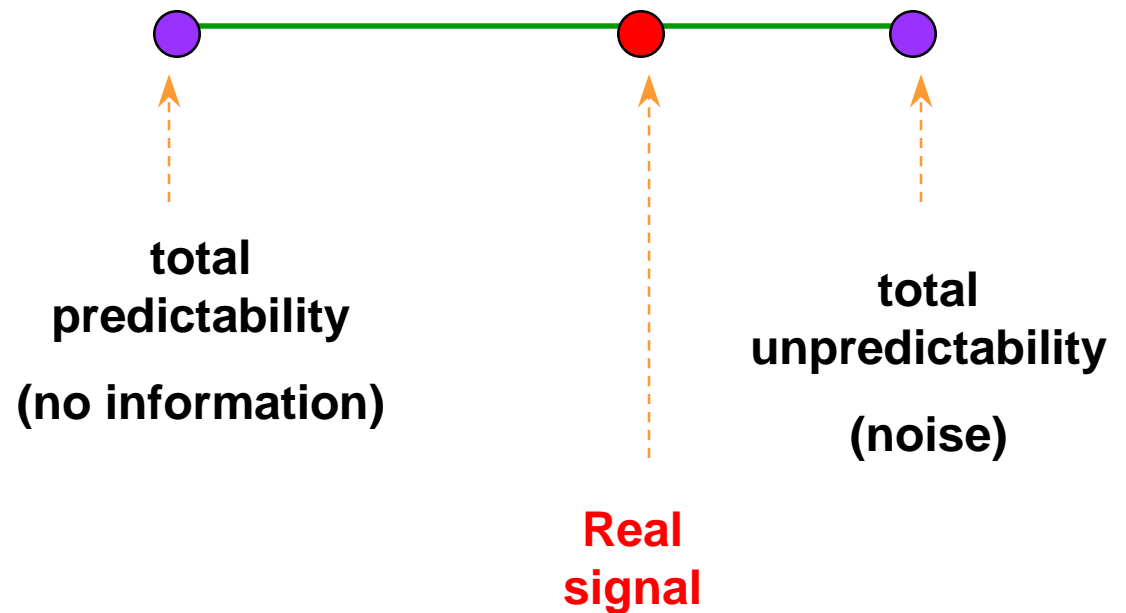
Information rate, bit rate, entropy

- Information rate is different from bit rate
- Information content or entropy of a sample:
 - a function of how different it is from the predicted value
- **Shannon's theory:**
 - any signal which is **totally predictable**: carries **no information**
 - (e.g. a sine wave)
 - **noise** is completely **unpredictable**: high entropy

Real, contentless signals, noise

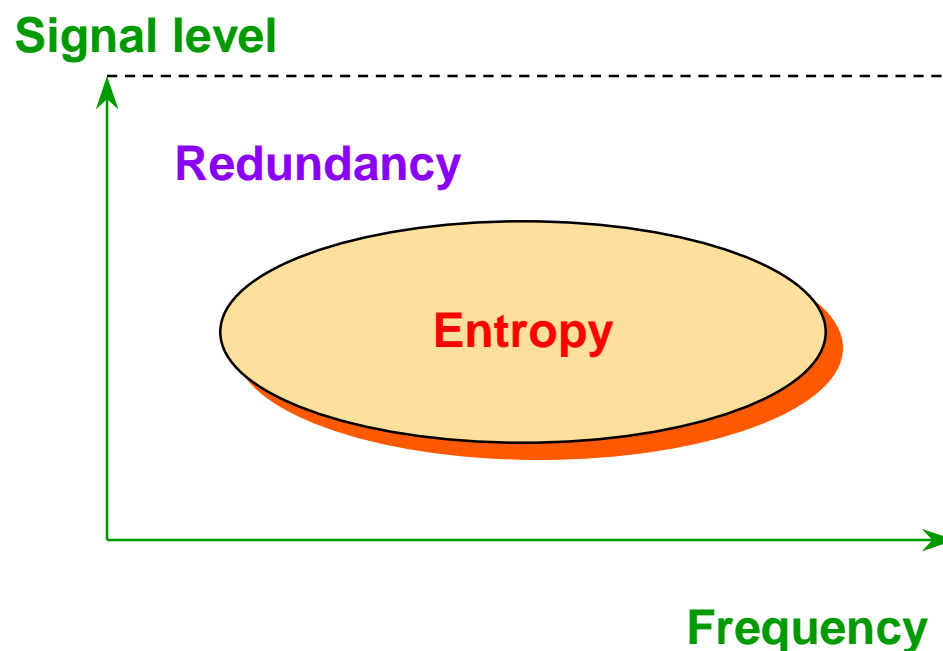
Real signals

lie somewhere
between the two
extremes



Entropy, redundancy, compression

- Real signals only occupy a portion of the Signal level - Frequency plane
- **Entropy** area: where the information lies
- **Redundancy** area adds no information



Compression: removes (part of) the redundancy

The Effect of compression

Compression removes redundancy ... *but*

Principle (or platitude)

Redundancy

**is essential for resistance to
errors**

The Effect of compression

Compression removes redundancy

- **Redundancy essential for resistance to errors**
- **Compressed data more sensitive to errors**

End of

Part 1

Requirements of Multimedia over the Internet

Part 2

Transporting Multimedia over the Internet

*Who has a good or fair understanding
of the difference between ...*

FTP *and* **HTTP**

HTTP *and* **TCP**

TCP *and* **UDP**

UDP *and* **RTP**

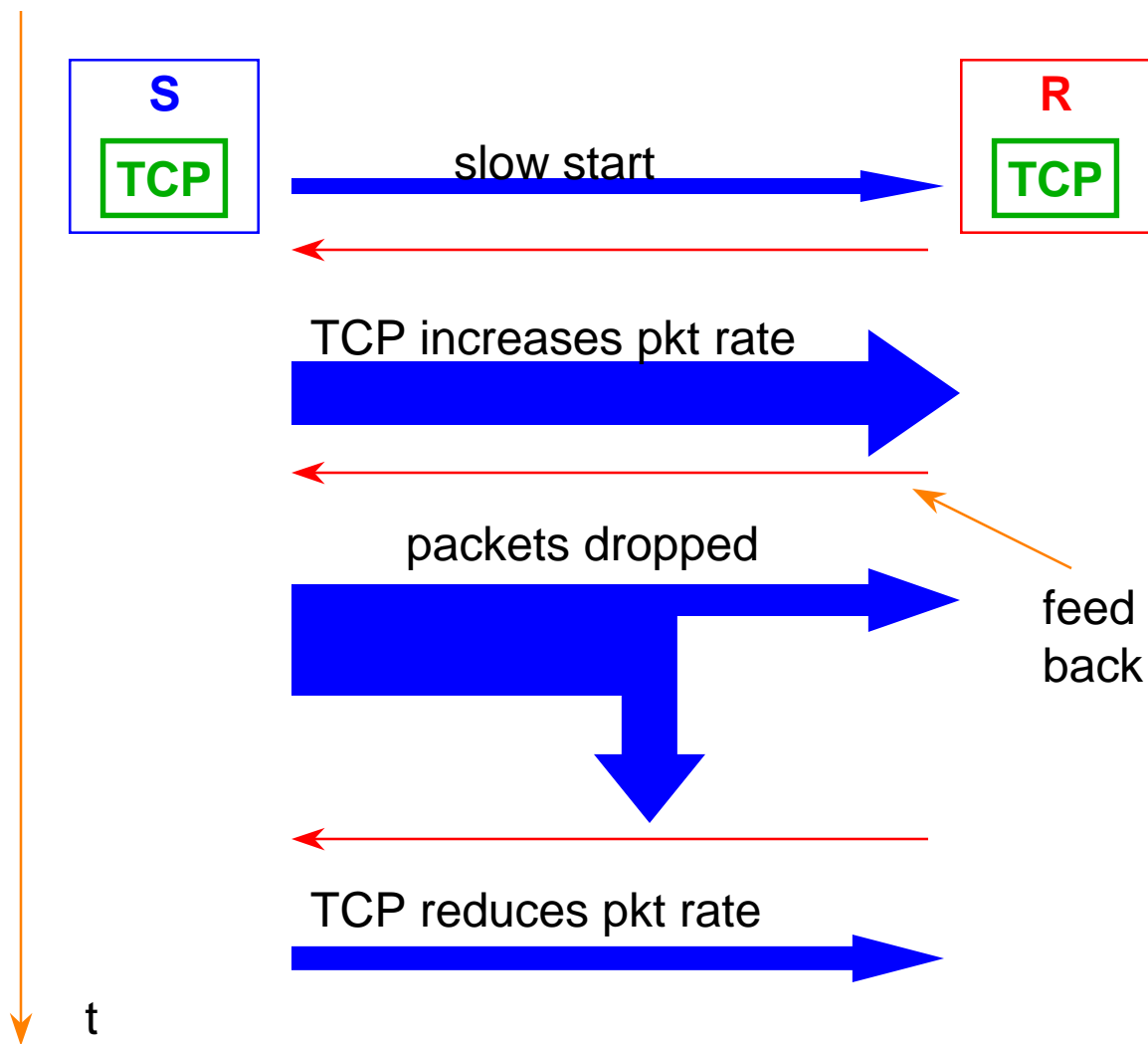
RTP *and* **RTCP**

TCP behaviour

- **Slow start**

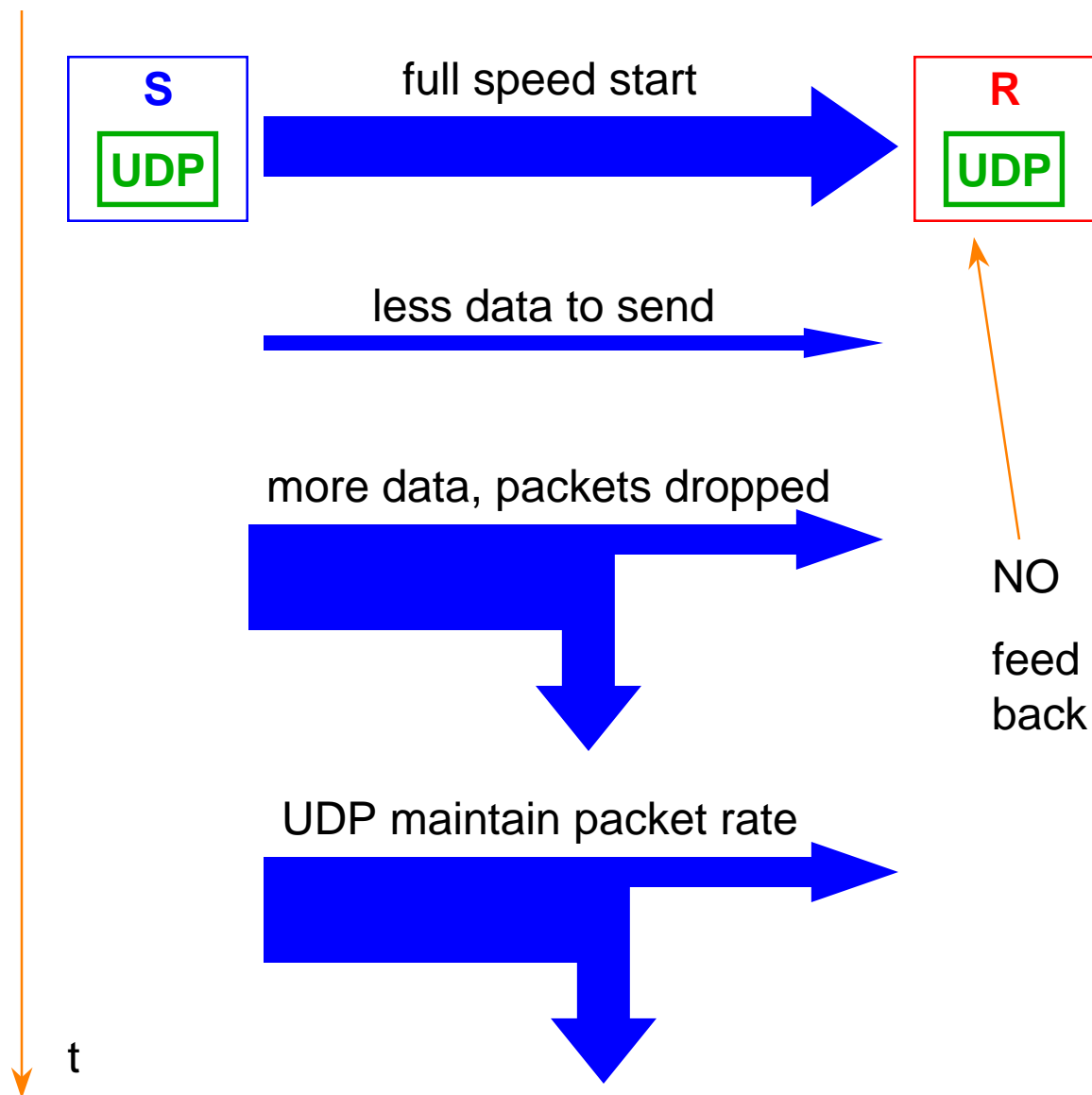
- **Sender aware of packets dropped**

- **Sender decreases bit rate when packet dropped**



UDP behaviour

- UDP sends blindly to a receiver
- **No feedback** from the receiver
- Sender unaware whether packets are dropped/lost



Protocols for real-time audio and video

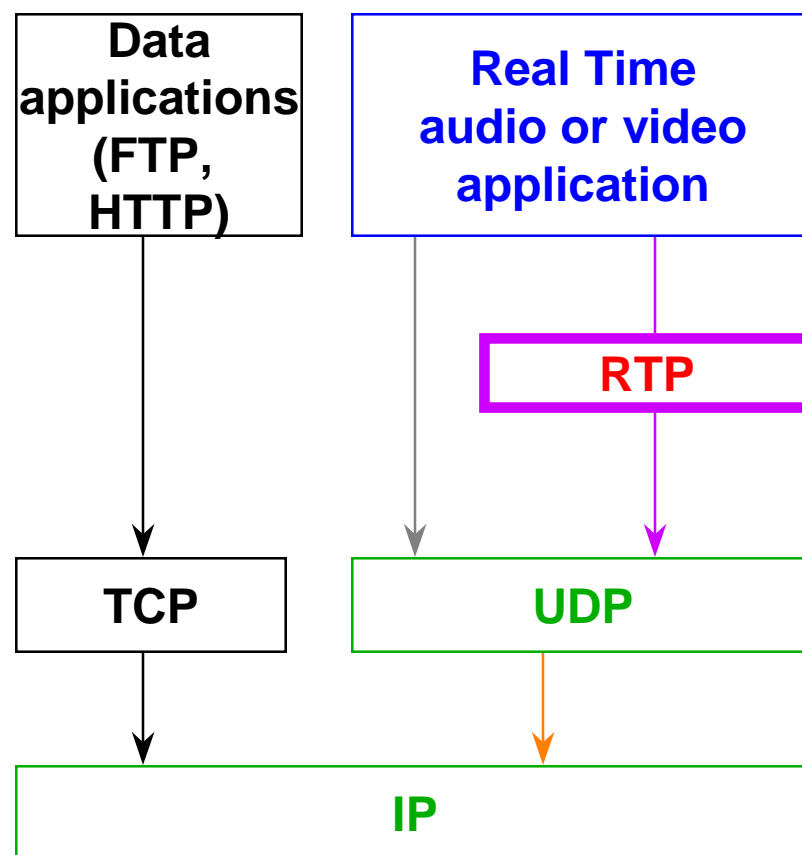
- Audio/video applications cannot operate over TCP

- They use UDP

- which has no timestamp, feedback, ...

- All applications use **RTP** (Real-Time Transport Protocol)

- time-stamp
- packet loss detection



Real-Time Protocol

- **RTP: an Internet IETF standard**

- **Supports**
 - timing reconstruction: timestamp (4 bytes)
 - loss detection: sequence number (2 bytes)

- **Lighter than TCP**
 - no retransmission, no flow control
 - TCP header: **20 bytes**; RTP header: **12 bytes**

Real-Time Protocol services

- **Two parts in RTP**

- **RTP** per se: for carrying data

- **RTCP**: to identify participants,
monitor the quality of the service

- **Session control (RTCP)**

- Receivers send periodically “reports”

- “Reports” indicate how good the reception is

- **RTP**

- Needs and Principles



- **Header overhead**

- **End-systems improvements**

- Redundancy coding

- Error concealment

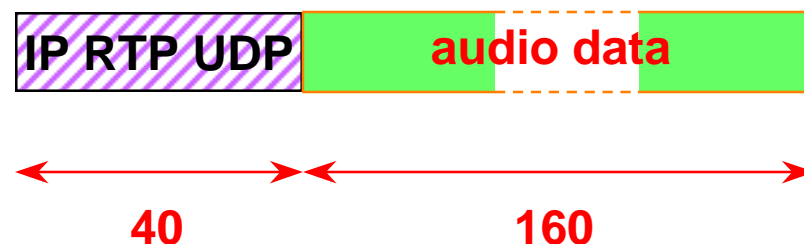
- **Quality of Service**

- **Unfair competition ...**

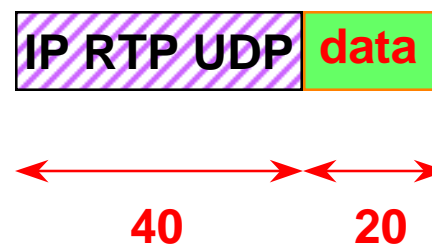
On header overhead

- IP+UDP+RTP headers = **40 bytes**

- At 64 Kbps PCM
 - 20 ms = 160 Bytes
 - overall rate = 80 Kbps



- At 8 Kbps (e.g. with G.729)
(e.g. over modem lines)
 - 20 ms = 20 Bytes



IP, UDP, RTP compression

- IP/UDP/RTP compression specified by
 - Robust Header Compression (ROHC) IETF draft
- Can reduce to **1 byte** (best case)
- Operates on a **link-by-link** basis

Basic principles

- **Fixed fields removal**
 - parts of the headers remain unchanged between pkts

- **Differential encoding**
 - some fields vary in a predictive, monotonic way

- **Re-coding combinations of fields**
 - some fields may be combined and hash coded

- **RTP**

- Needs and Principles

- Header overhead



- **End-systems improvements**

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- **Quality of Service**

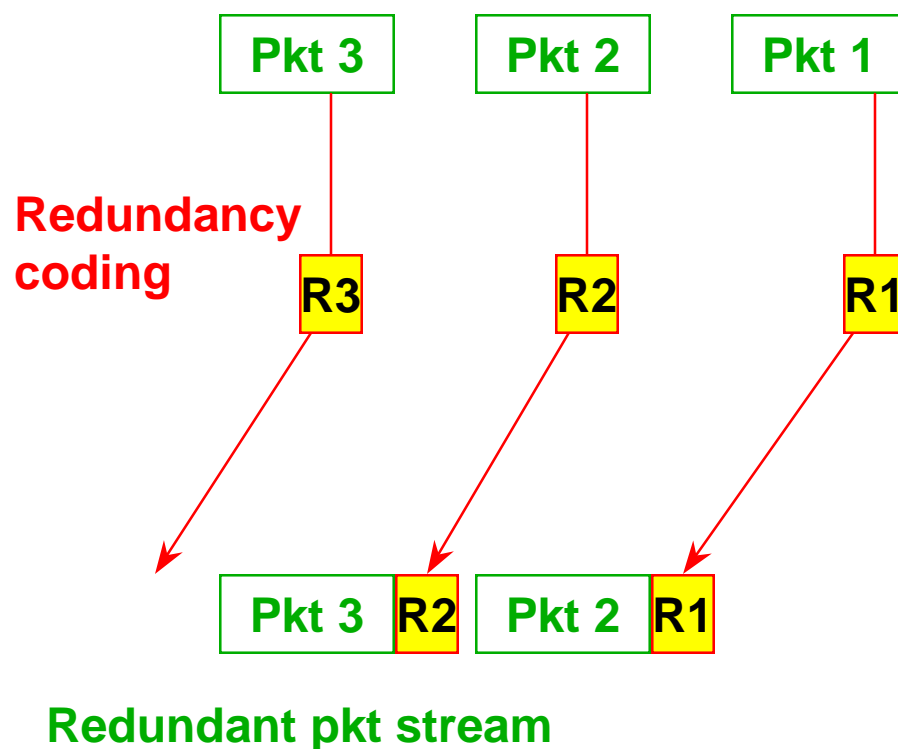
- **Unfair competition ...**

Low-bit rate redundancy

Compression aims at removing redundancies ... but redundancies improve resistance to data errors

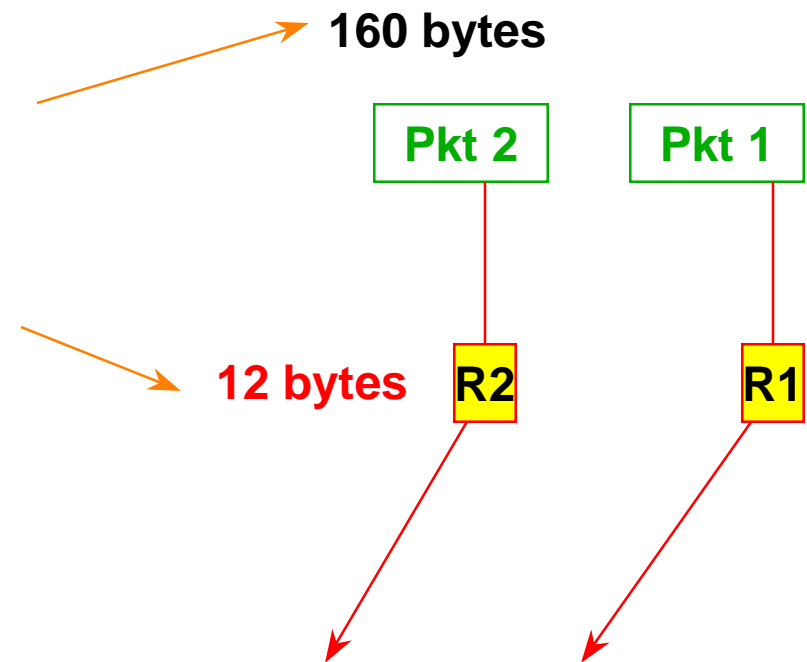
- re-code each packet at lower resolution
- insert re-coded packet into one subsequent pkt(s)

Primary coding



Example of redundancy: RAT (UCL)

- Primary coding = 64 Kbps
- Redundancy coding = 4.8 Kbps
- Experiment shows reasonable repair with high loss rate (40%)



- **RTP**
 - Needs and Principles
 - Header overhead
- **End-systems improvements**
 - Redundancy coding
 - **Error concealment**
- **Quality of Service**
- **Unfair competition ...**



Error concealment (audio example)

Replace missing packet with

- **silence**

- “OK” if $\text{pkt} < 16\text{ms}$, loss rate $< 1\%$; beyond, clipping effect (1)

- **white noise**

- (better than silence)

(1) “OK” means tolerable; does not mean unnoticed

Phonemic Restoration

- brain uses phonemic restoration:

“the ability of the brain to subconsciously repair a missing segment of speech with the correct sound”

- phonemic restoration
 - occurs better when missing segment replaced by **noise** instead of silence

- **RTP**
 - Needs and Principles
 - Header overhead
- **End-systems improvements**
 - Redundancy coding
 - Error concealment
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Improving QoS

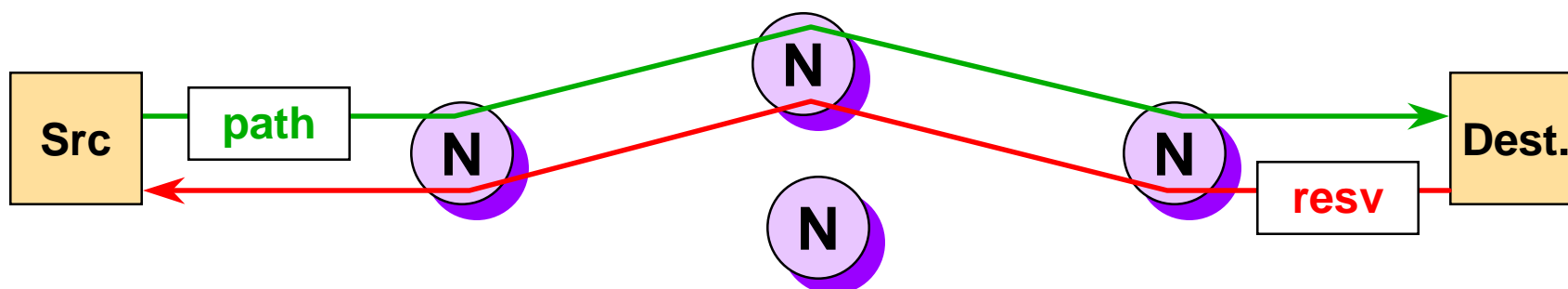
Combination of solutions

- **Integrated Services** : RSVP Protocol
- **Differentiated Services**

IntServ principles

- **Resource reservation** is necessary
- Reservations on a **per-flow** basis
- Routers have to maintain **flow-specific states**

RSVP protocol (simplified)



- “path” control message sent periodically by source
- “path” establish an RSVP route in intermediary routers
- sink replies with a “resv” message, according to its capabilities
- “resv” reserve resources in node on the route back
- if “path” not repeated after time-out, resources released
- “path” and “resv” are carried by ordinary best-effort datagrams

Concern: Scalability

■ Problem

- how many soft-states can network handle at a time?
- problem of **granularity** of the flows

RSVP scalability

- Millions per-flow RSVP reservations in high speed backbone?

- *Problem 1:*

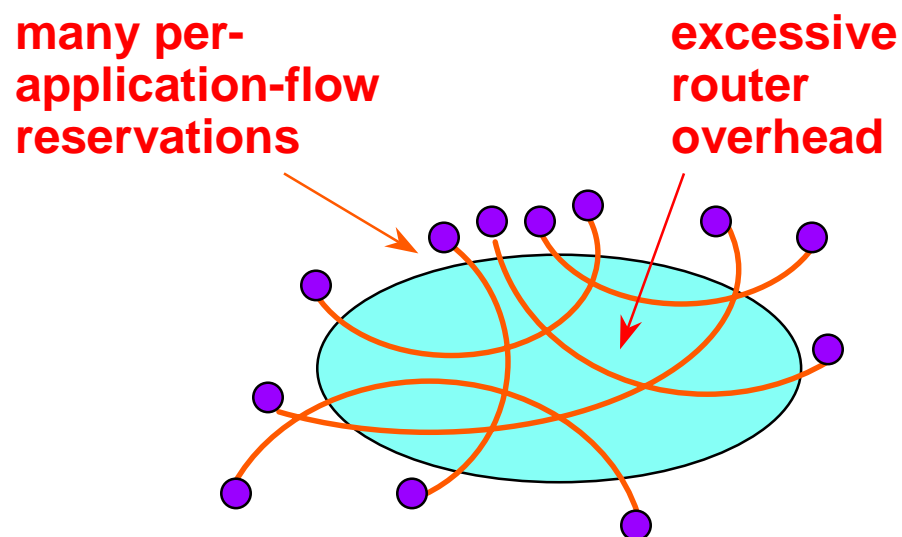
Signaling overhead

- CPU: PATH/RESV processing
- memory: states

- *Problem 2:*

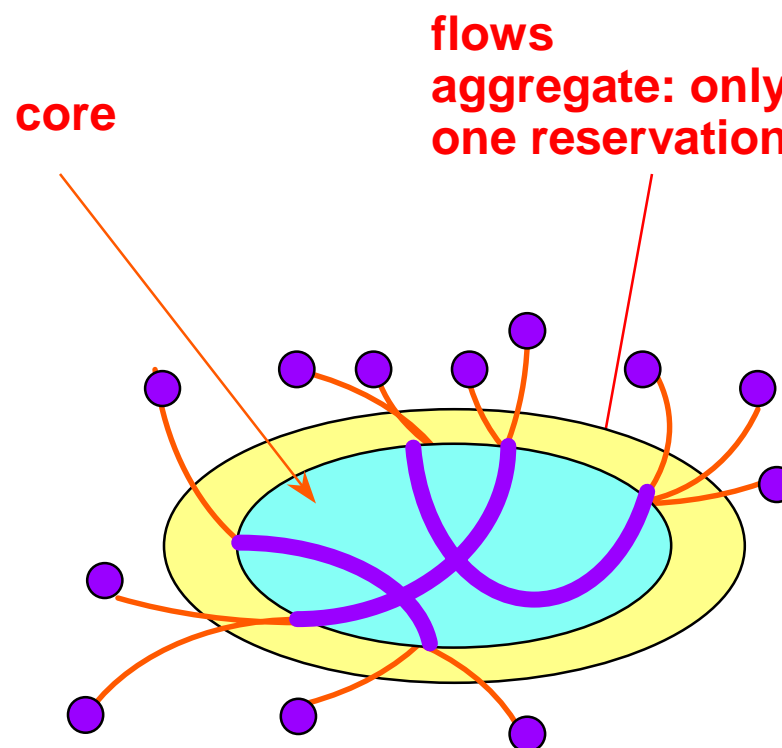
Data pkt overhead:

- CPU: Multi-field classification



Solution to *problem 1* (Signaling Overhead)

- **RSVP aggregation** in core
- Only one reservation between ingress/egress pairs or routers
 - If N boundary routers,
 - N^2-N RSVP reservations
- Size of the aggregate does not need change for every new flow request

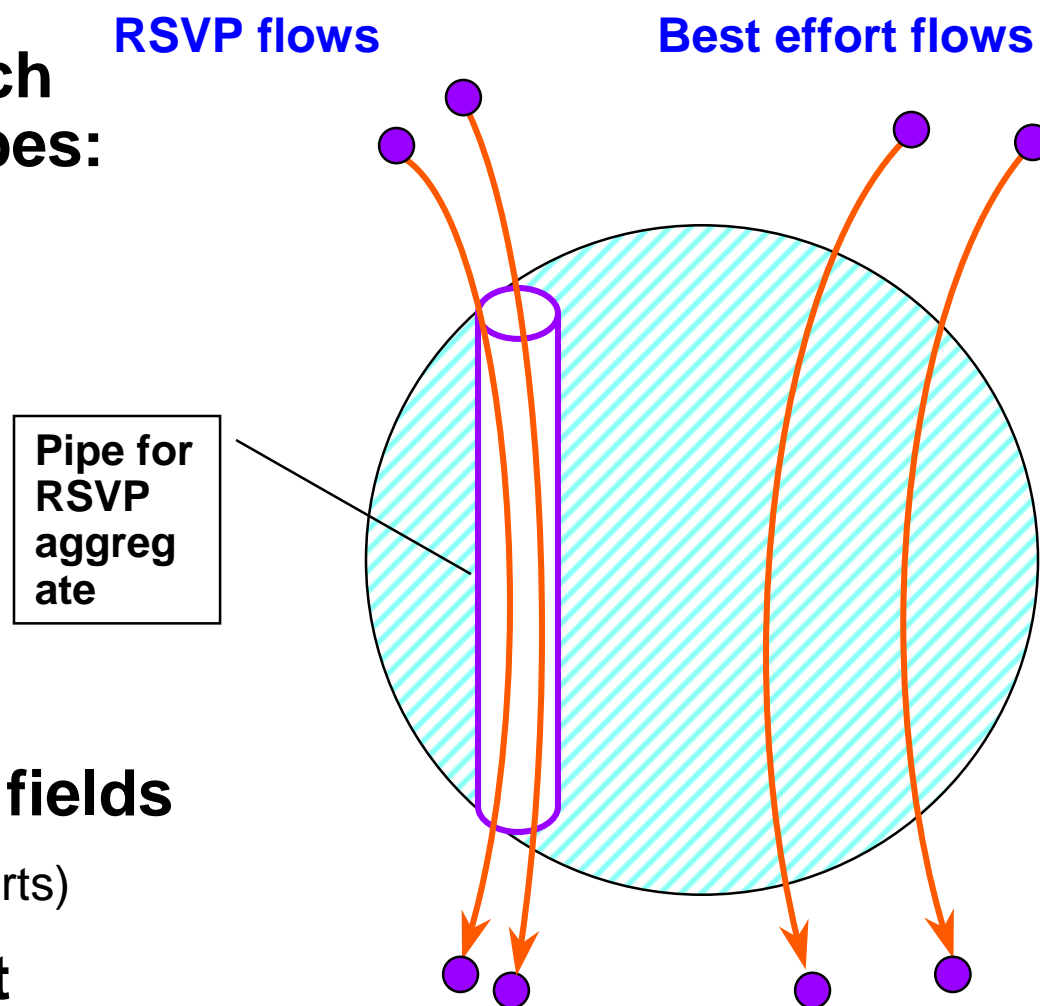


Problem 2: overheard of classification

- Recognizing pkts which belong to reserved pipes:

heavy process!

- Requires
 - examining multiple fields
(source, dest addresses, ports)
 - match them against
“filters”



Differentiated Services : principles

■ Rationale

- Knowing pkt priority needs heavy classification process
- Classification may be faster if packets are “marked” (1) with a priority

(1) “Marking” pkts also called “coloring”

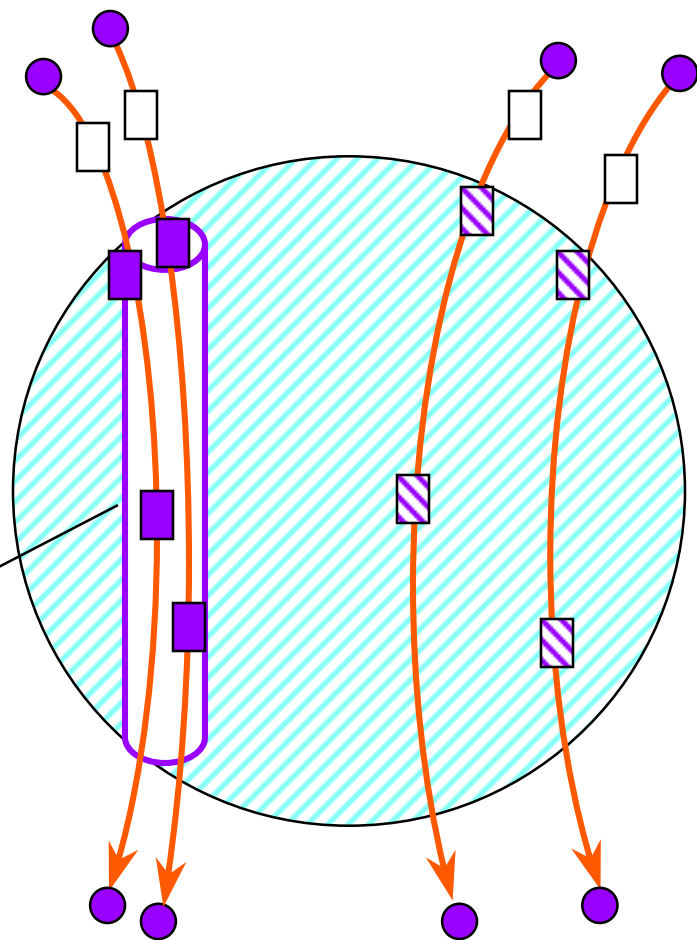
Differentiated Services Packet marking

- Pkts carry explicitly their priority

■ High Priority
▨ Low Priority

RSVP flows Best effort flows

Pipe for RSVP aggregate



Principle (or platitude)

- Systems with no reservation (e.g. connectionless networks) **scale well**, but are **poor at QoS** guarantees

Too bad for IP, Ethernet

- Systems with reservations (e.g. connection-oriented networks) are **good at QoS** guarantees and **poor at scaling**

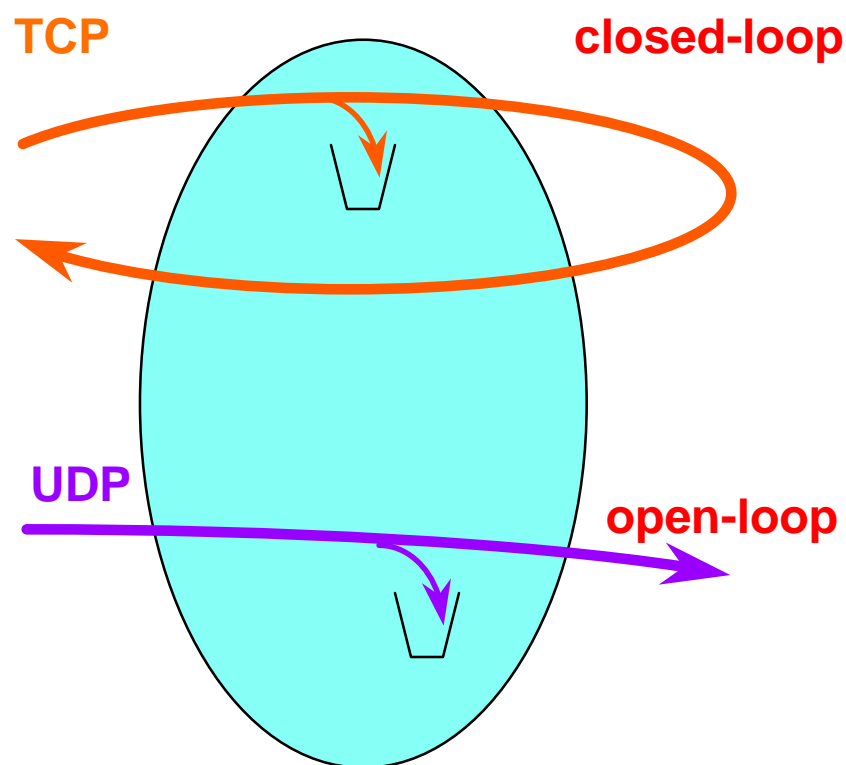
Too bad for RSVP, ATM

- **RTP**
 - Needs and Principles
 - Header overhead
- **End-systems improvements**
 - Redundancy coding
 - Error concealment
- **Quality of Service**
- **Unfair competition ...**



Unresponsive flows

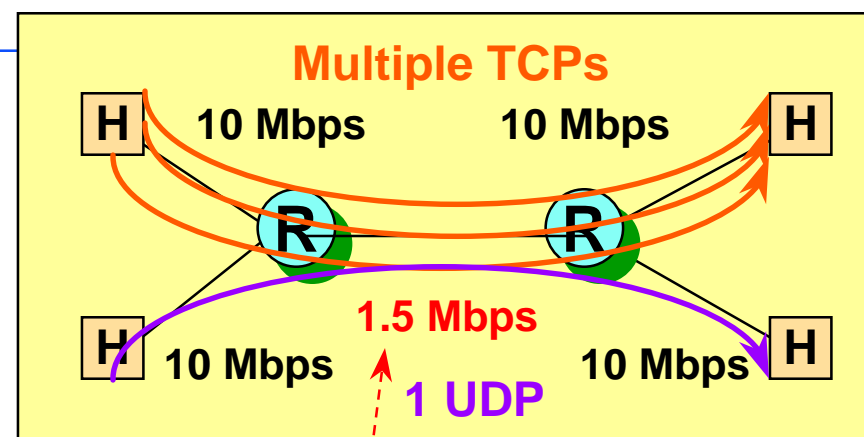
- Unresponsive flows do not react to congestion indication (pkt loss)
- Can create
 - bandwidth **starvation** inflicted to well-behaved responsive traffic
 - Congestion **collapse** (network busy transmitting pkts that will never reach dest.)



Unfair competition

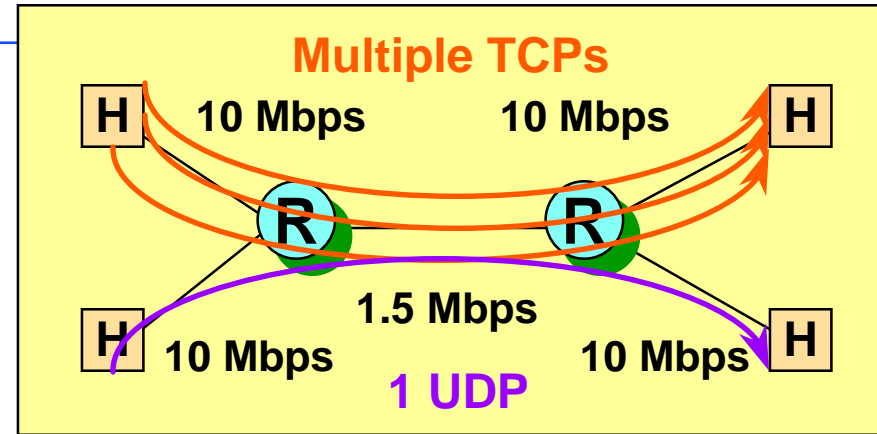
- **Case 1:**
 - **2 LANs (10 Mbps)**
 - **interconnected with T1 and a pair of routers**

- **Competition between**
 - **3 TCP connections and**
 - **1 UDP connection**

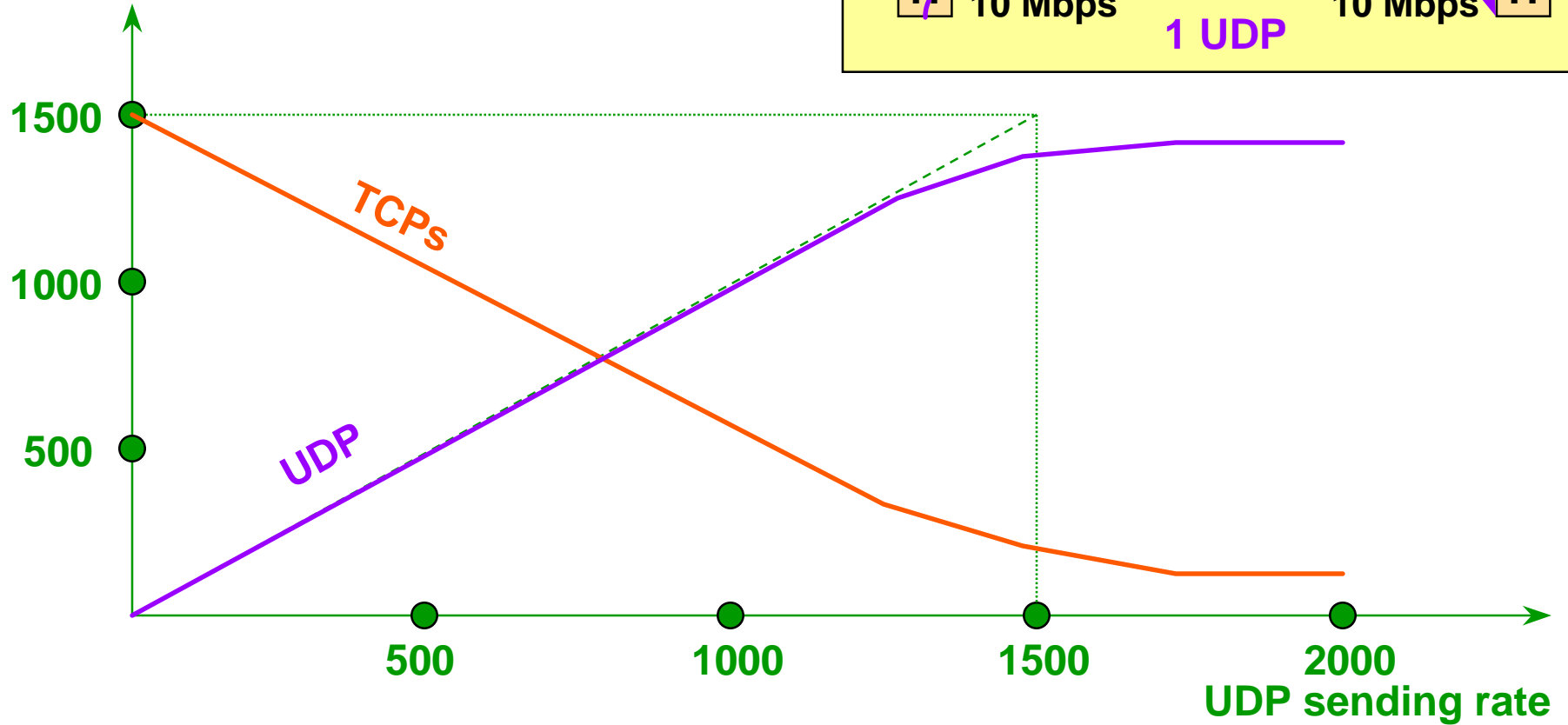


Bottleneck

Unfair competition



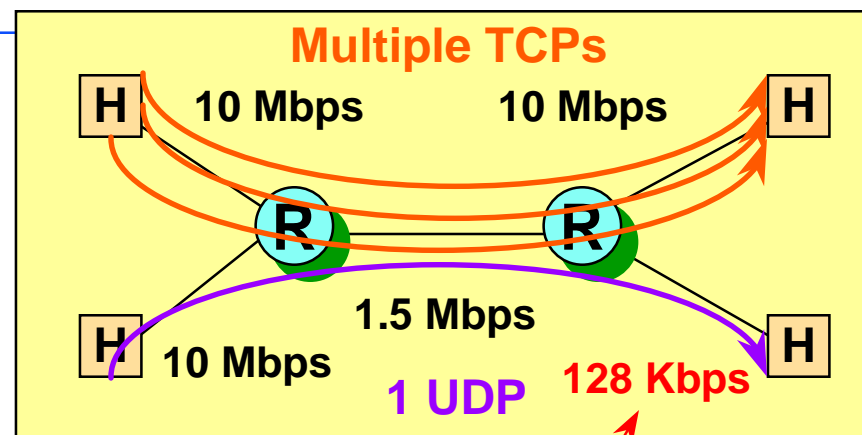
Delivered bandwidth ("goodput")



Congestion Collapse

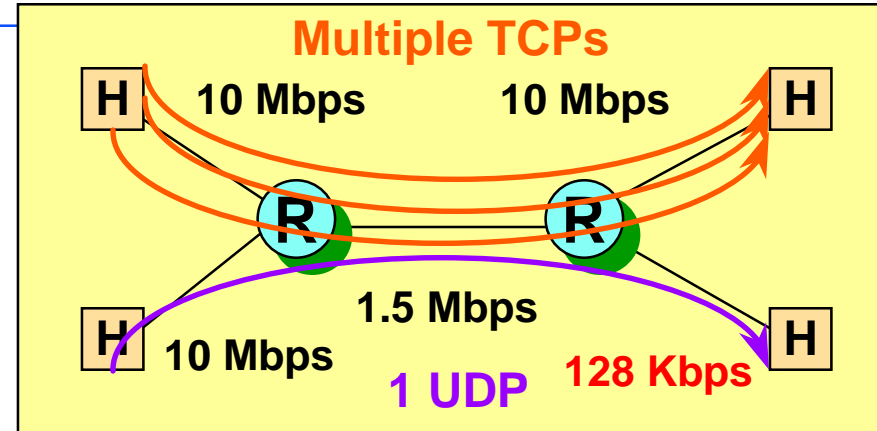
- **Case 2:**
 - **2 LANs (10 Mbps)**
 - **interconnected with T1 and a pair of routers**
 - **UDP receiver connected via ISDN (dual)**

- **Competition between**
 - **3 TCP connections and**
 - **1 UDP connection**

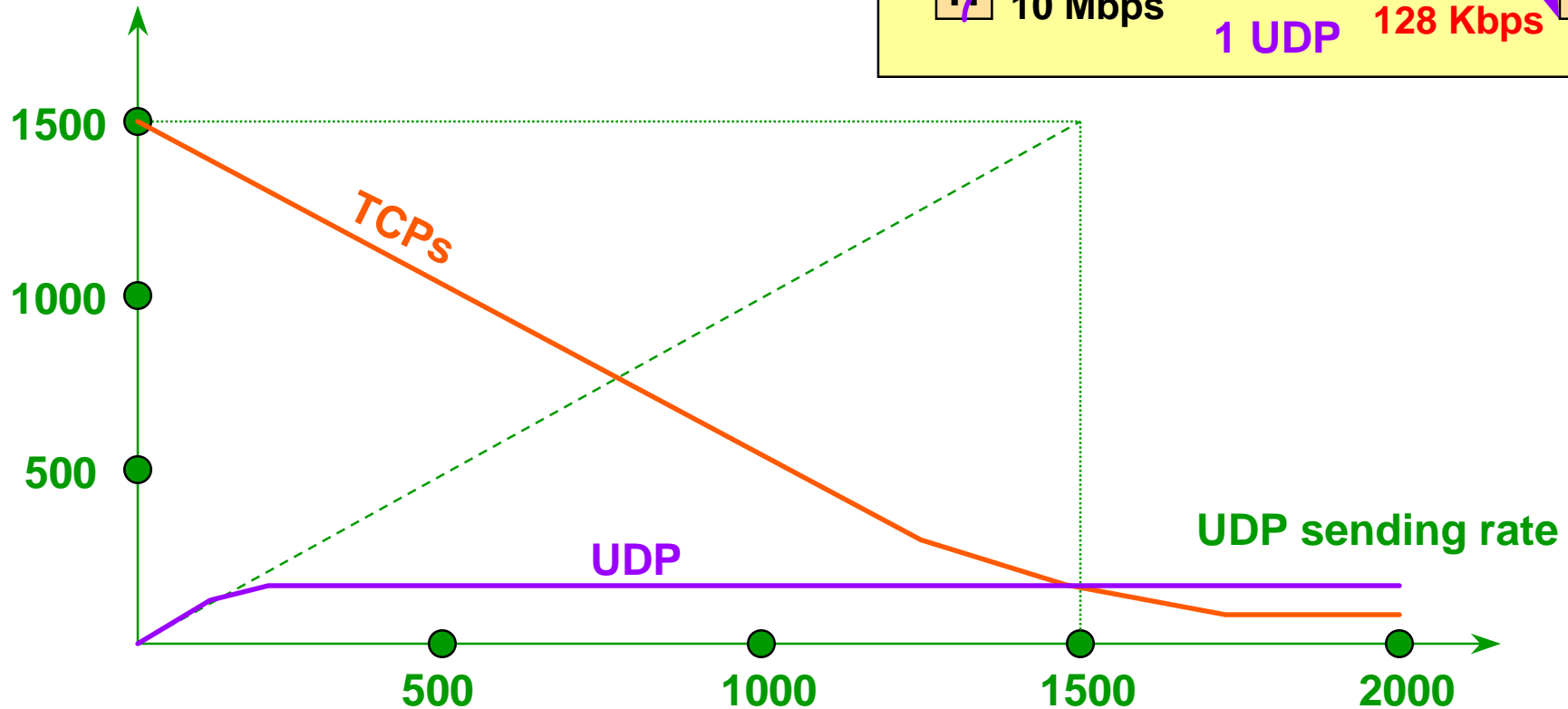


Bottleneck

Congestion Collapse



Delivered bandwidth
("goodput")



Future or non-TCP applications

- **In the future, UDP traffic may be penalized in public Internets**
- **Real-time Applications will have to behave "more responsively" to congestion indications**
 - adaptive compression levels
 - hierarchical encoding/compression
 - frame, block rate reduction for video
 - ...

Future or non-TCP applications

- **In the future, UDP traffic may be penalized in public Internets**
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 - adaptive compression levels
 - hierarchical encoding/compression
 - frame, block rate reduction for video
 - ...

Epilog

*“640KB should be enough
for anybody”*

Bill Gates, 1981

*“We’ll have infinite
bandwidth in a
decade’s time.”*

Bill Gates, 1994

Predictions is a difficult art,
in particular when ...

... predicting **the future**

Principle (or platitude)

When you have
no memory of the **past**,

you cannot
predict the future

Where the future is?

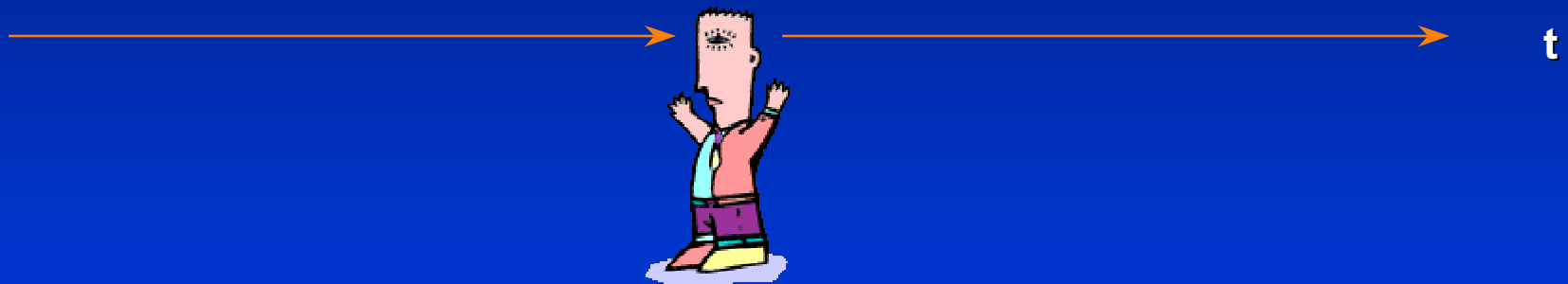
- **Dr Raphael Nunez**

- **Prof. Cognitive Psychology and Ethno-Mathematics,
Uni of Friburg, Swizerland**

*Studied how a people in South America see
mentally “the future”*

- **Spatial Localization of the future becomes a
phenomenon of “naturalization”**

Where the future is?





Thank you