Mohammad Kheider University of Biskra
Faculty of Exact Sciences and Natural and Life Sciences
Computer Science department

Master 1 course
Option: RTIC

QUALITY OF SERVICE (QOS)

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Course Map

- **Multimedia communicating systems**

  - Principles and architectures

  - Streaming systems and ToIP

  - RTP/RTCP, SIP, RTSP protocols

- **Quality of service**

  - Principles and mechanisms

  - Classification, scheduling, queue management, congestion control, admission control, routing with QoS

  - Protocols IntServ,DiffServ

1. Multimedia communicating systems

Definitions

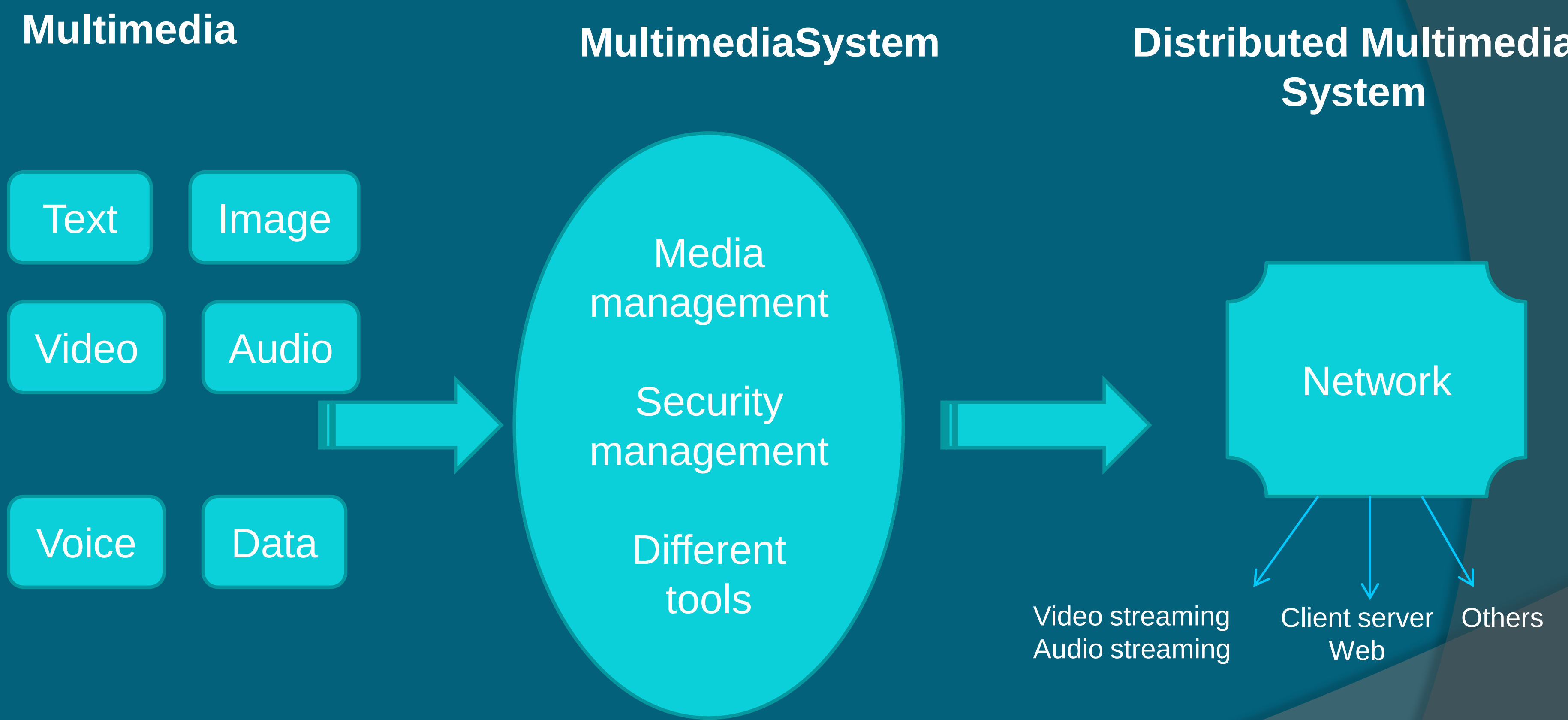
- **Multi: Indicates multiple**

- **Multimedia (computing): presentation of an application (generally interactive) which integrates elements such as text, graphics, video, sound.**

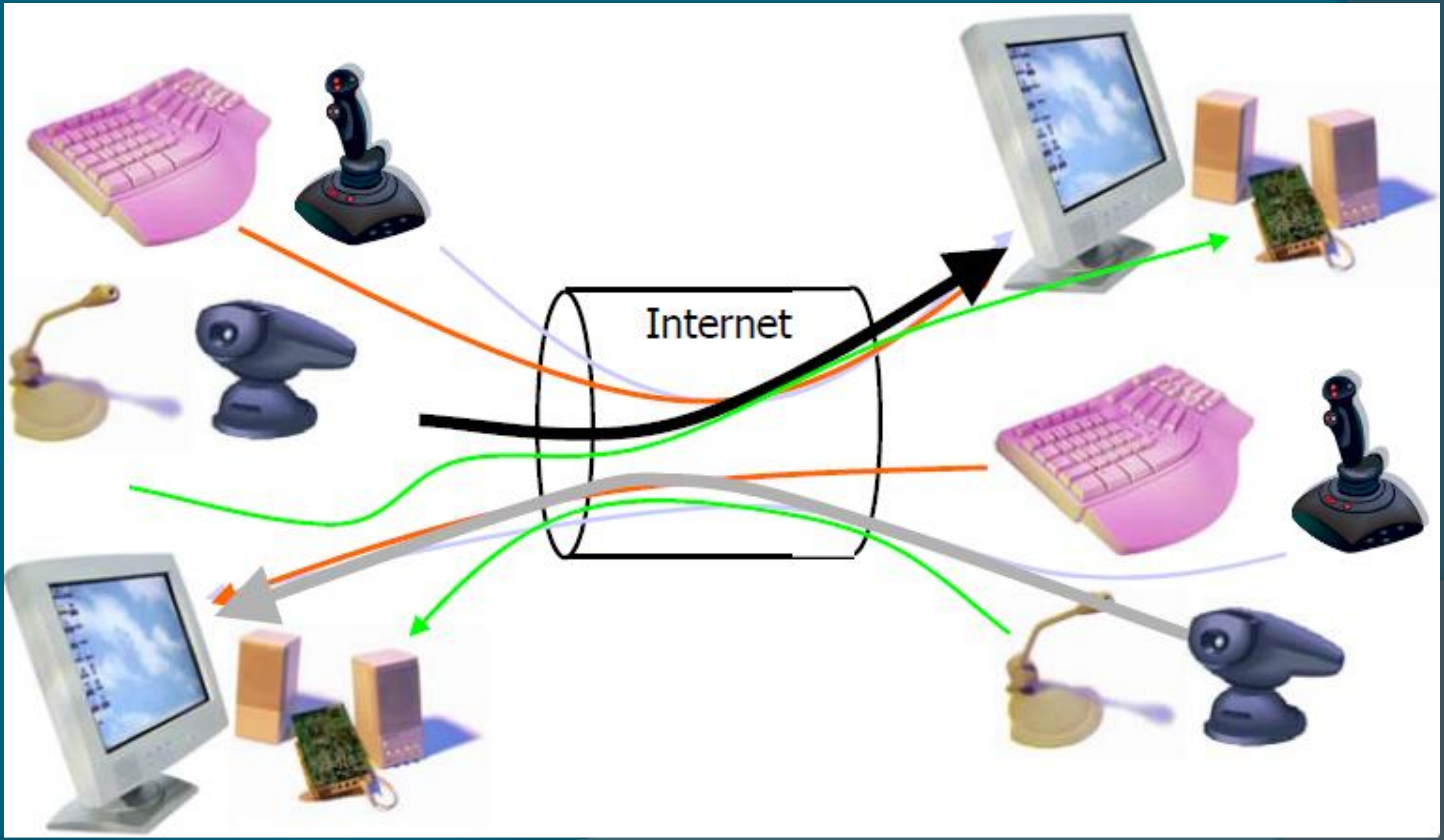
- **Media: means/support for the dissemination, distribution or transmission of signals carrying written, sound, visual messages (press, cinema, radio, TV, etc.)**

- **Multimedia system: Computer and associated software used to run a multimedia application.**
- **Distributed multimedia system: an SMD that operates on a set of equipment interconnected by a communication network.**

Concepts related to *Multimedia*

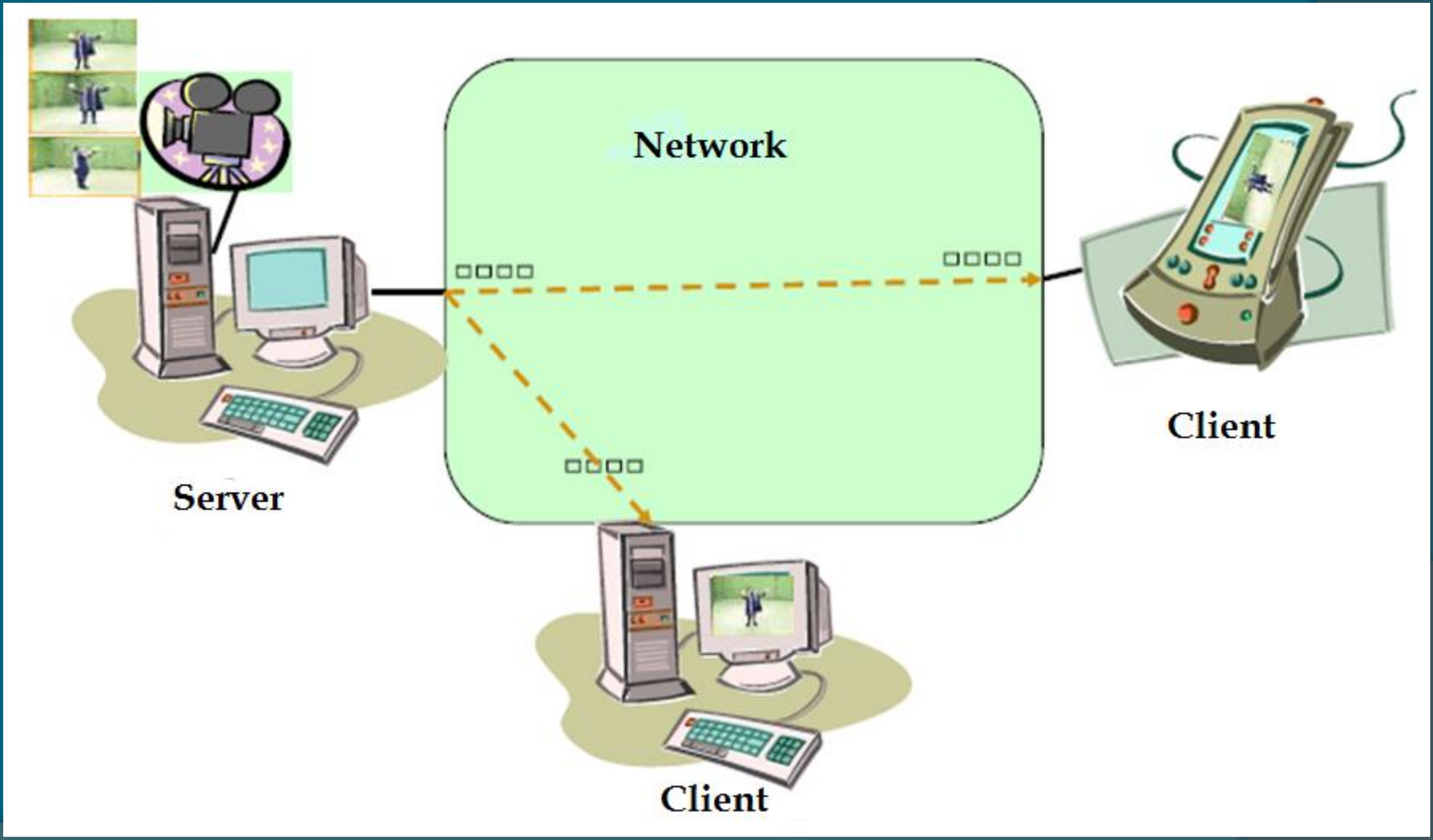


Wired equipment for *Multimedia*



Wireless and mobile equipment for the *Multimedia*

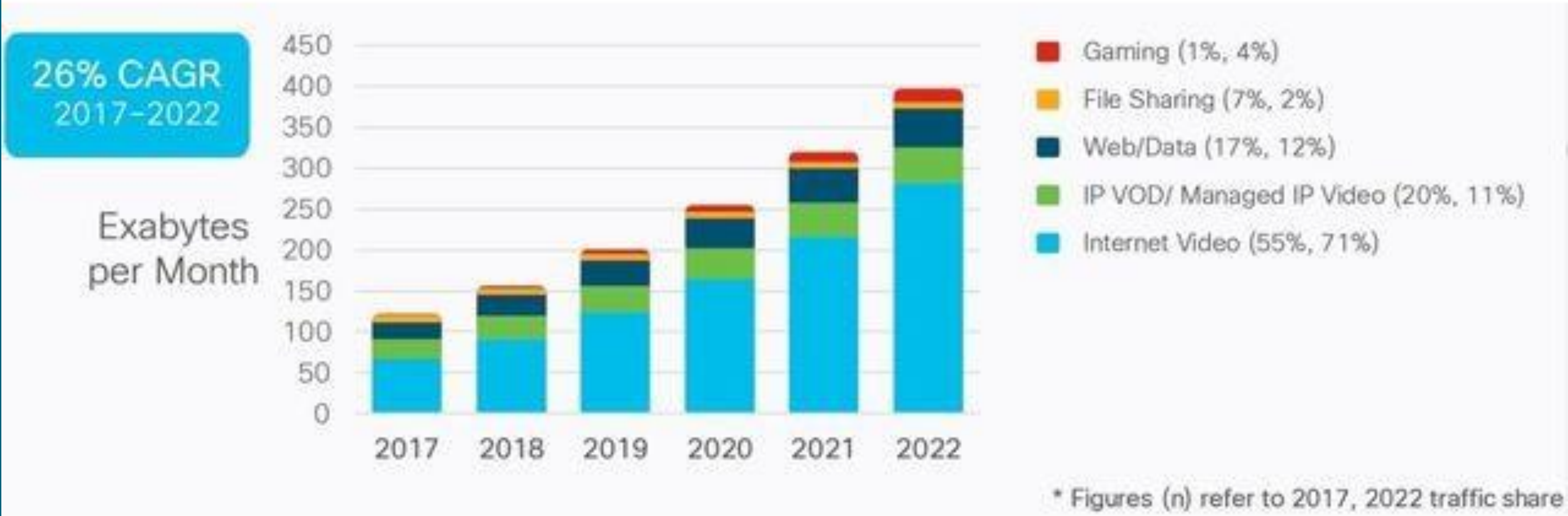
General principle of distributed multimedia applications



Classification of multimedia applications

- **According to the interactivity**
  – Non-interactive: radio and TV, video on demand, e-learning...
  – Interactive: video surveillance, remote control, video conference call,telemedecine,teleshopping,games…
- **According to the criticality**
  – (Very) critical: guidance and supervision, telesurgical operation…
  – Average criticality : video conference, stock market, teleshopping
  – Non-critical: TV, radio, games…
- **According to the timings (real time)**
  -Sstreaming of previously stored audio/video data
  –Real-time 1-to-m streaming of audio-video data
  – Interactive audio/video applications

Evolution of Internet traffic



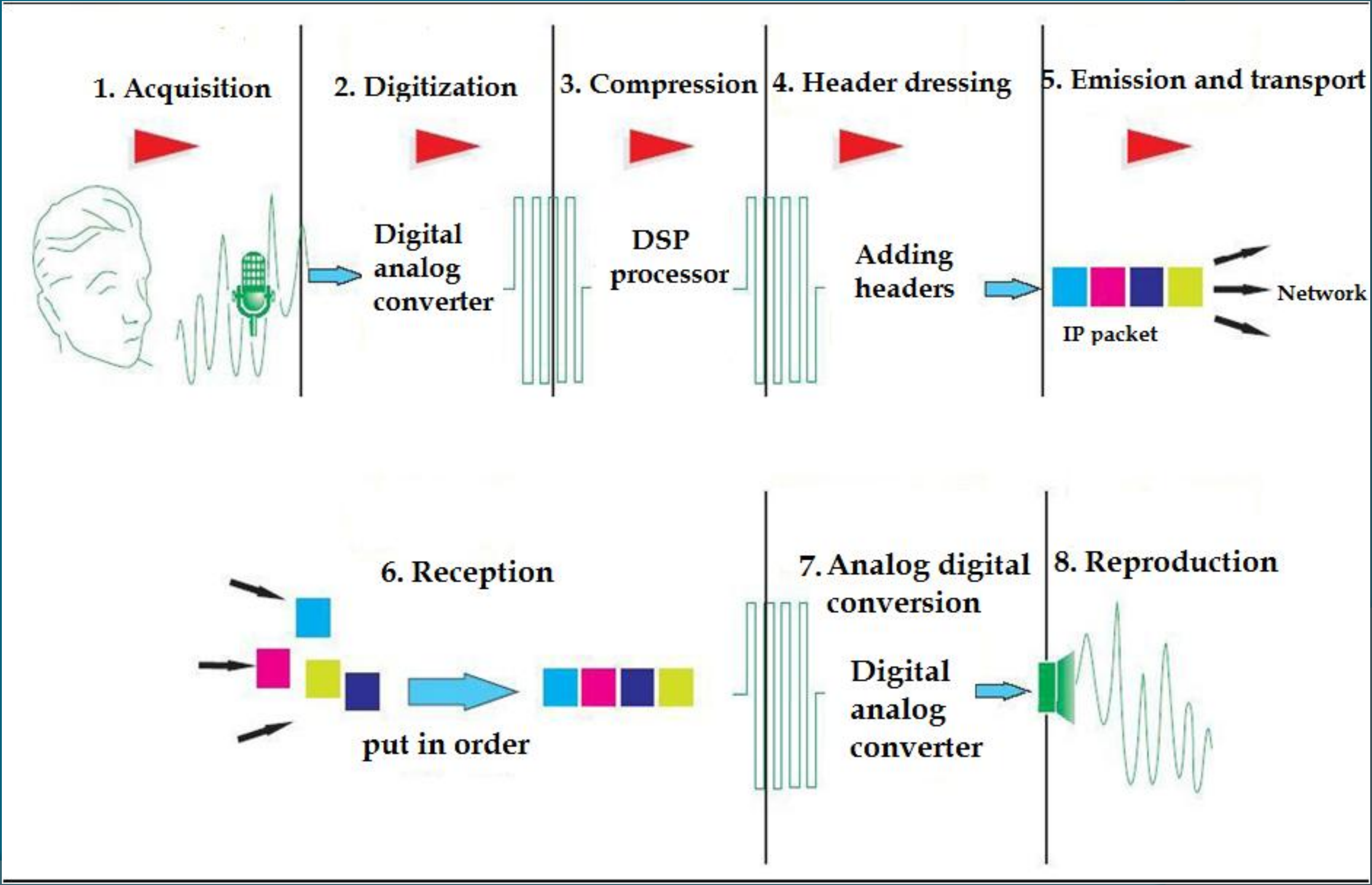
1Exabytes= 1018bytes

Source: Cisco VNI Global IPTraffic

Digitization and compression

- Audio/video support not necessarily digital
  – Digitize content
- Scanned data may be large in size
  – Compress
  – Codecs
- Compression/Decompression
- Choice of codec often imposed by network bandwidth
  – Lossless compression
  – Lossy compression

Principles



1. Acquisition
2. Digitization
3. Compression
4. Header dressing
5. Emission and transport
Digital analog converter
DSP processor
Adding headers
IP packet
Network
6. Reception
put in order
7. Analog digital conversion
Digital analog converter
8. Reproduction

Audio digitisation

- Pulse Code Modulation – PCM

- Compression techniques

- Voice
  - GSM (13 kb/s), G.729 (8 kb/s), G.723 (6.4 and 5.3 kb/s)
  - proprietary techniques

– CD quality music
  - MP3
  - 96, 128 and 160 kbps
  - splitting into independent files
  - Streaming
- Others: AAFC,Vorbis, …

Video digitization

- Video

– Sequence of images viewed at a certain bit rate

- Picture

– Suite of pixels

- pixels

– Luminance and color

– Encoded in a number of bits

Video compression

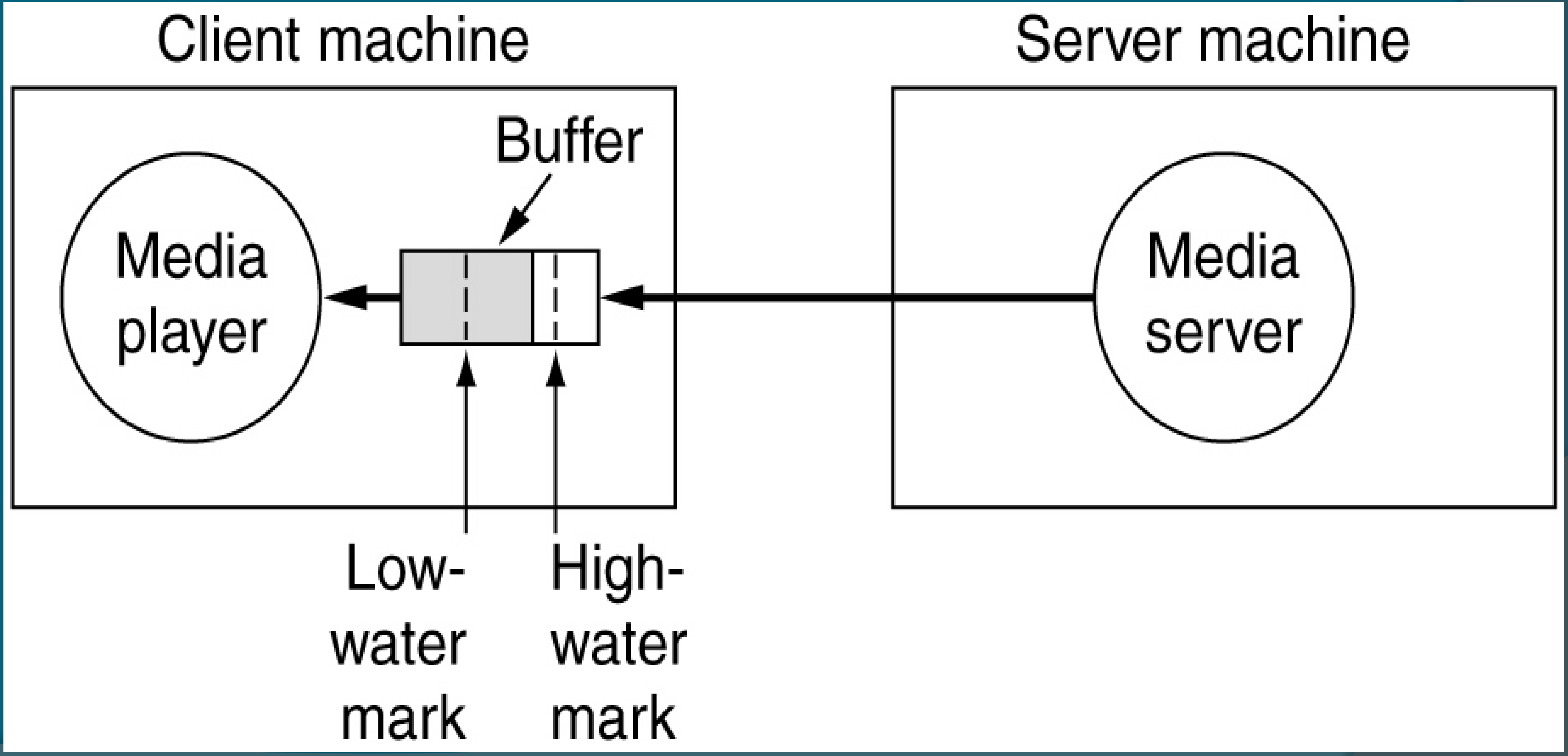
- Redundancies
  - Spatial redundancy
  - Temporal redundancy
- MPEG compression standards
  - MPEG 1 CD-ROM video quality – 1.5 Mb/s
  - MPEG 2 high quality DVD video – Digital TV – 3-6 Mb/s
  - MPEG 4 tt type of multimedia applications
  - Inspired by the JPEG standard
- Other standards
  - H.261, 262, 263, 264
  - Owners

Audio/video streaming

- Definitions
- Play an audio/video stream as it is broadcast
  - No need to have downloaded the whole file
  - The download continues in the background
  - Temporary data storage
  - Alternative to download
- Stored
  - The requested file is previously stored on a server
  - eg. video on demand
- Real time / live
  - Similarity to broadcast radio/television
  - Real-time content processing and delivery

Audio/video streaming

- The “media” client softwareplayer» puts the data in a buffer, and then plays it



Telephony over IP :ToIP

- Differences betweenVoIPand theToIP

• Voice over IP

– Transmit an audio signal in the IP world

• Telephony over IP

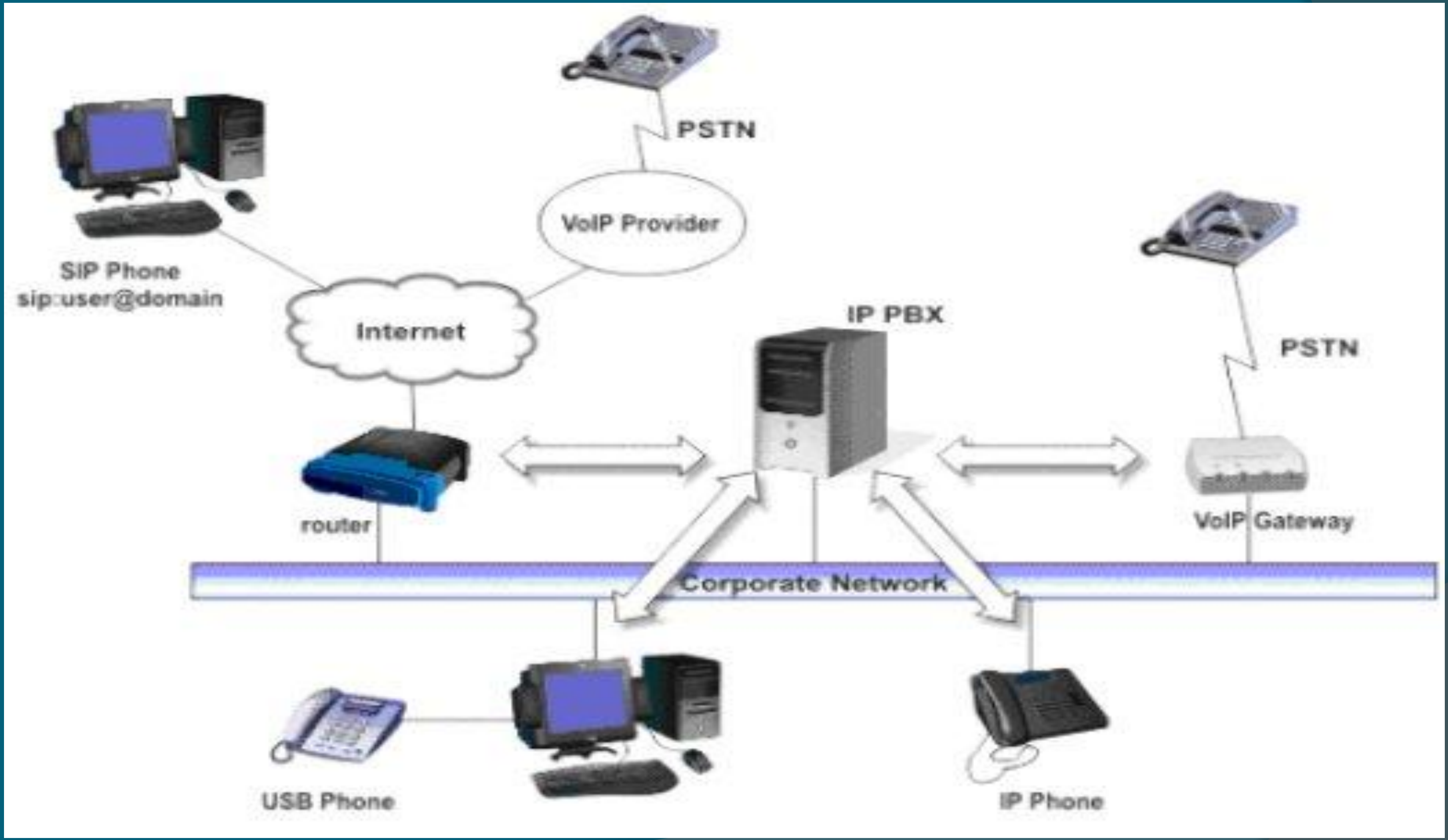
– Application of theVoIP

– Telephone functions and services around theVoIP

• which allow telephony

– IP telephony architecture

Differences between VoIP and the ToIP



Benefits of ToIP

- Users

- Cost

- Long distances

– Flexibility

- IP phone mobility
- Physical and material mobility
- Operators

– No strong regulation

– Management of a single network

- Voice – data
- Cost
- 60% of thebpallocated to a voice circuit (PSTN) not used

VOIP protocols

- The main protocols used for establishing connections in voice over IP are:

Equipment forToIP

- Telephones

–Softphones

- Software to be installed on a system computer science

–Hardphones

- Conventional telephones with a socket ethernet

- Configuration files

Equipment forToIP

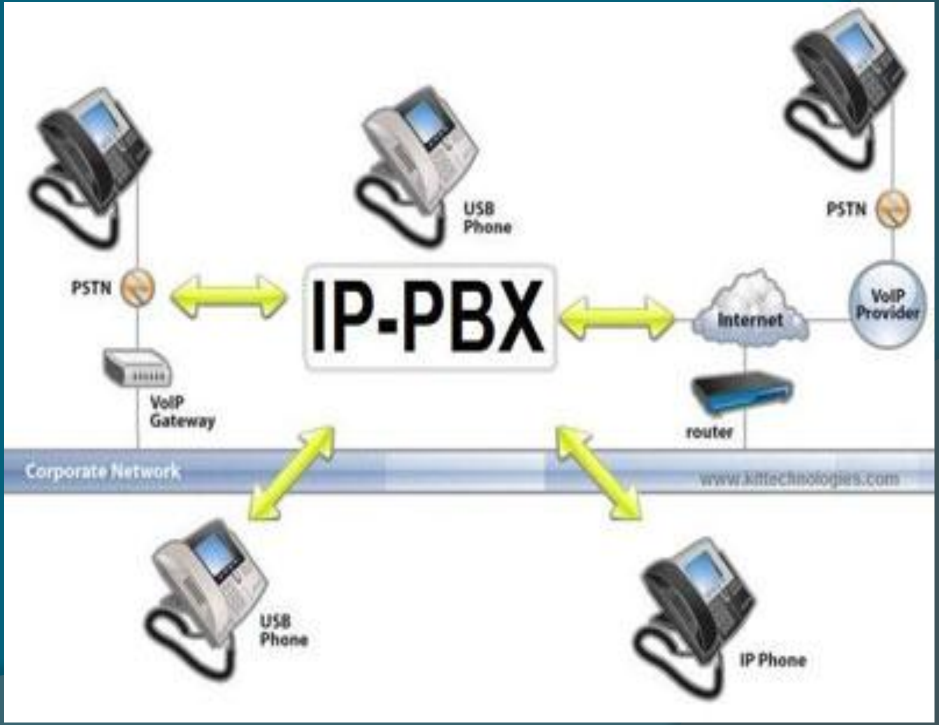
- PBXs

– IP-PBX (PABX –Private Automatic Branch exchange)

– Management and interconnection post offices

– Provision of services telephone

– Hardware / software



Principle of IP telephony

- Telephone communication: talk, silence, talk...

– Normally: it takes 64 kb/s during the speech phase

- Packets are generated only during speech phases

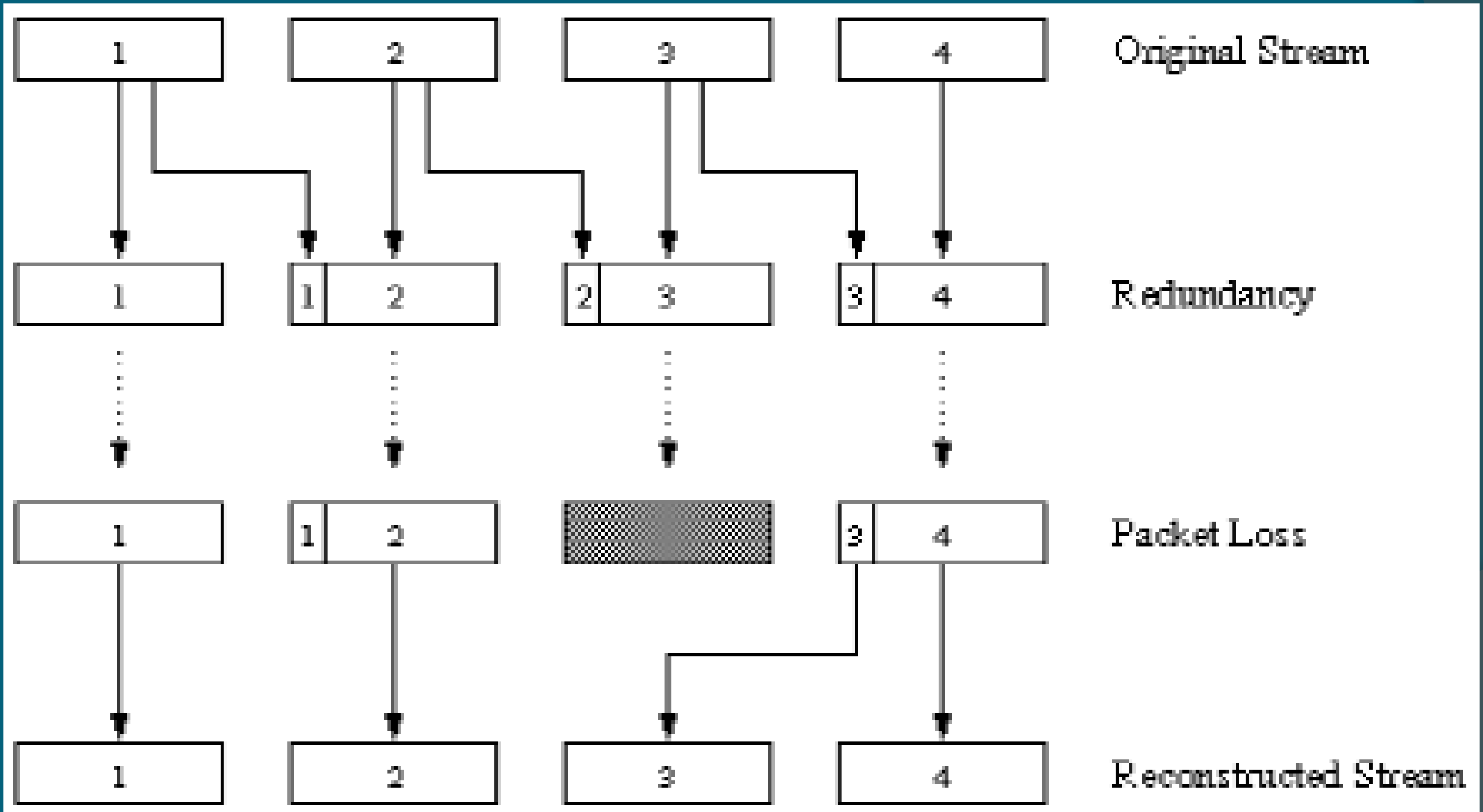
– Message = a piece of speech (of 160 bytes of data) + header

- Each message is encapsulated in a UDP segment.
- The application sends UDP segments via the UDP-socket every 20 ms during the speech phases. The sending rate is 8 kb/s.
- Up to 10% (or even 20%) packet loss is tolerable.
- Packets with a delay greater than 400 ms are discarded upon receipt.
- Jitter is managed by using packet timestamps, sequence numbers, and by delaying certain packets before they are listened to by the receiver.

Packet Loss Recovery

- As retransmissions are inappropriate in a real-time context, an overlay strategy must be put in place. In the case of IP telephony, two techniques are used to reduce the impact of losses: FEC (*Forward errorcorrection) and Interlacing.*
- **Recovery by FEC**: Add redundancy information by mixing the values of several pieces in a packet.

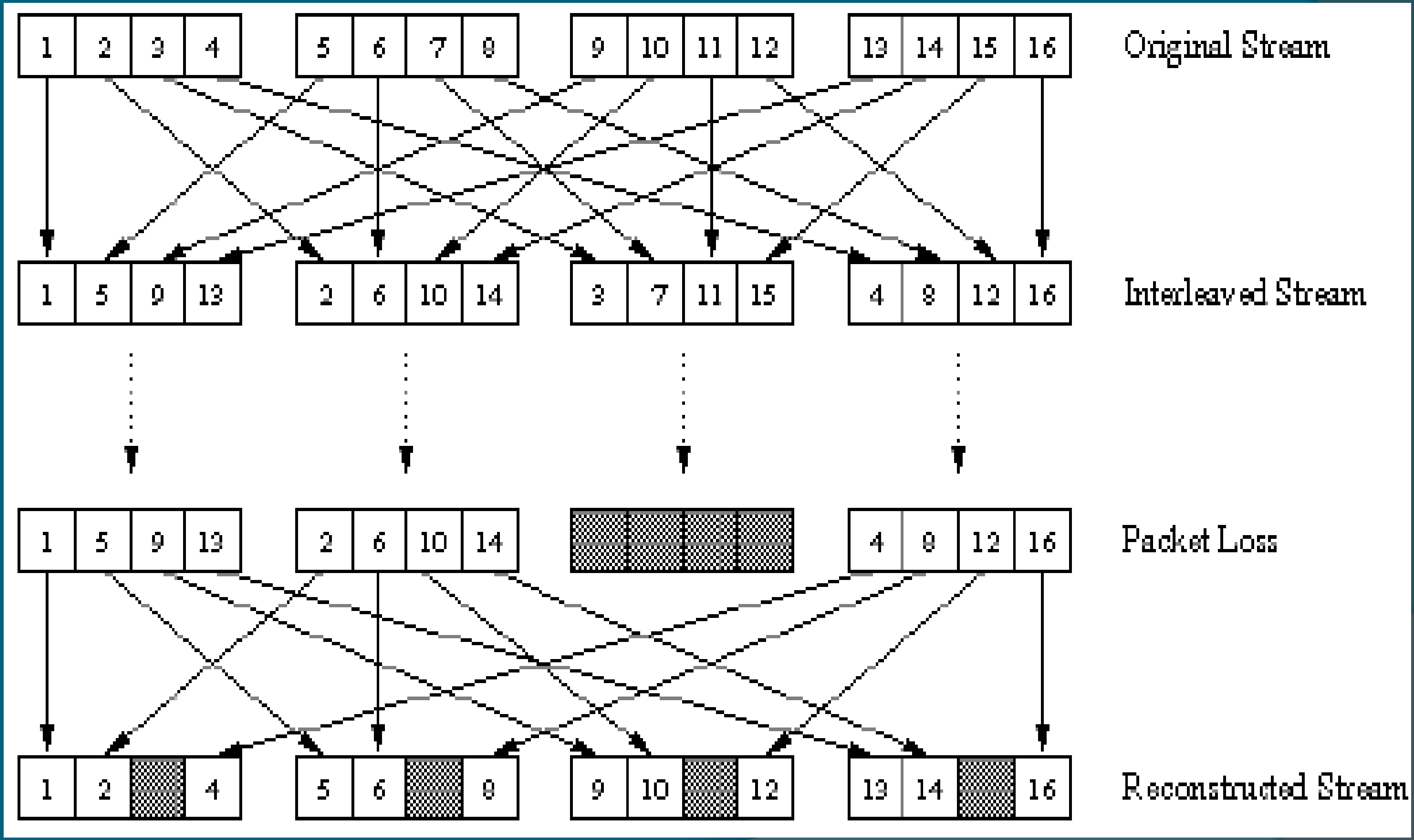
Recovery by FEC



Overlapping by interlacing

- No redundancy, but it may cause delays in the playout.
- Divide 20 msec periods of speech into smaller 5 ms periods and interleave the smaller chunks
- If lost, use incomplete pieces (rather than losing large whole pieces).

Overlapping by interlacing



Characteristics of multimedia applications

- Handling large amounts of 'continuous' data
- Minimum flow rates are required
- Delivery of information respecting timings
- Interactive applications require low round trip times
- Coexistence (and resource sharing) with non-media applications
- Resources required:

– Processors (high performance)
– Powerful servers
– Dedicated main memory (for buffering by the customer)
– Large capacity disk memory
– Network bandwidth with minimal latency

Multimedia application requirements

- Requirements: delay, jitter, throughput

■ The required values change with the evolution of the technological offer:

We do not ask the same things for a 56 kb/s Internet connection as for a 10 Mb/s connection.

■ The (human) user knows both how to be demanding and how to adapt to what is offered to him.

■ Current demand trend: ever shorter lead times, ever higher throughputs, ever lower loss rates.

Multimedia application requirements

■ Telephony and audio conferencing

– Low throughput (~ 64 Kb/s), but delays should be short (< 250 ms)

■ Video on demand

– High throughput (~10 Mb/s), non-critical latency

■ Video conferencing

– High throughput for each participant (~1.5 Mb/s), low delay (< 100 ms), synchronized states.

■ Distributed music rehearsal

– High throughput (~1.4 Mb/s), very low latency (< 100 ms), high media synchronization (drift between sound and image < 50 ms)

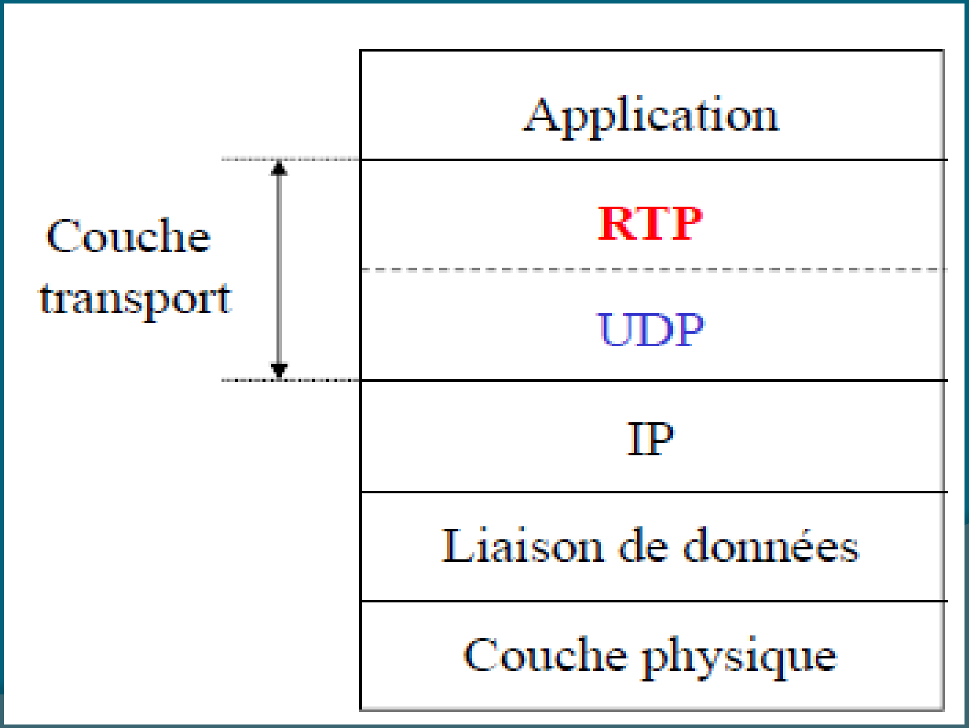
■ Games

– A maximum delay of 70 ms is more appreciated by gamers than a delay of 200 ms.

– The jitter should be 20 ms maximum, because the player adapts his strategy to a fixed delay (by shooting the targets for example). High jitter leads to boring gameplay.

Protocols for data transport multimedia
Real-Time Protocol (RTP) (1/3)

- RTP: a solution for AMMs with Internet in best effort
- Basically works on top of UDP



Real-Time Protocol (RTP) (2/3)

Type de flux	Numéro de séquence	Estampille	Identificateur de source de synchronisation	Données

RTP packet header

– Type of stream (7 bits)

– Sequence number (16 bits): used to detect losses.

– Timestamp (32 bits): Provides the sampling instant of the first byte of the packet.

It is used to absorb jitter.

– Sync Source Identifier (32 bits): identifies the source of the stream.

Each stream in RTP has a source-assigned identifier randomly (but distinct from those that already exist) at the start of the stream.

Real-Time Protocol (RTP)(3/3)

Some types of audio streams supported by RTP

Stream type	Audio Format	Sampling	Rate
0	PCM	8 KHz	64 Kb/s
1	1016	8 KHz	4.8 Kb/s
3	GSM	8 KHz	13 Kb/s
7	LPC	8 KHz	2.4 Kb/s
9	G.722	8 KHz	48-64 K/ps
14	MPEG Audio	90 KHz	----
15	G.728	8 KHz	16 Kb/s

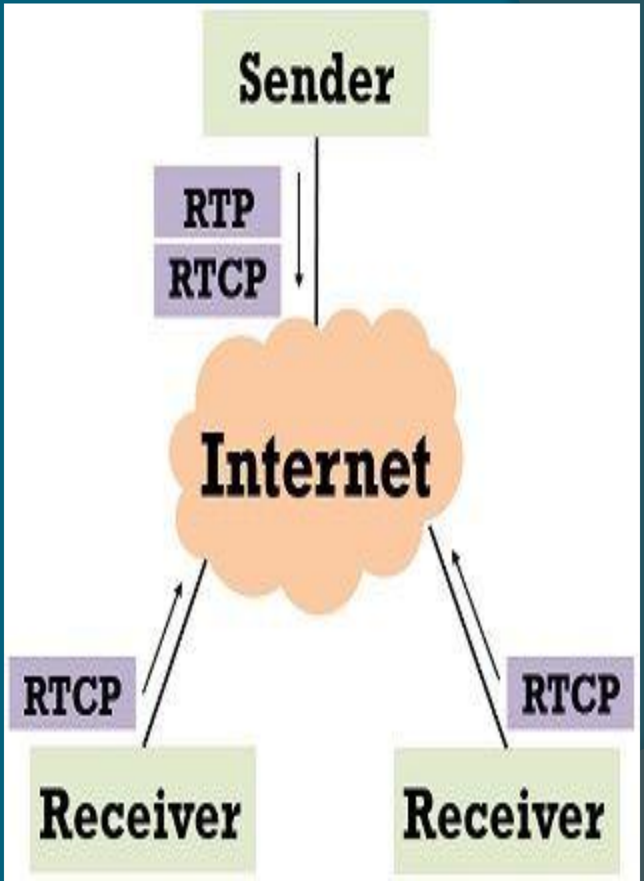
Some types of video streams supported by RTP

Stream type	Video Format
26	Motion JPEG
31	H.261
32	MPEG1 Video
33	MPEG2 Video

Real-Time Control Protocol (RTCP)(1/2)

- RTCP is used to route packets containing reports about a media stream between a source and a receiver.

- Reports contain statistics on: number of packets transmitted, number of packets lost, transfer jitter, etc.
- Report packets are sent by receivers, possibly at the request of sources.
- Report packets are used by the source to modify/adapt its timing to network conditions.



Real-Time Control Protocol (RTCP) (2/2)

- If each receiver sends its report packets to all the other sources/receivers of the stream: significant network overload.

- RTCP adjusts the time intervals between reports based on the number of receivers participating in a stream
- Typically, the bandwidth used for RTCP is limited to 5% of the session bandwidth. This fraction is shared between the report requests issued by the sources (25%) and the reports issued by the receivers (75%)

– *Ts: period of transmission of RTCP packet by the source:*

$$T_s = \frac{Nombre\ de\ sources}{5\%*25\%*Bande\_passante\_session} *Taille\_paquet\_moyen\_RTCP$$

– *Tr: RTCP packet transmission period by a receiver:*

$$T_r = \frac{Number\ de\ récepteurs}{5\%*75\%*Bande\_passante\_session} * Taille\_paquet\_moyen\_RTCP$$

Signaling Protocols: RTSP, SIP.

<u>RTSP</u>: Real-time streamingprotocol

• Client/server type application level protocol

• What it does not do

– Choice of compression techniques

– Choice of encapsulation

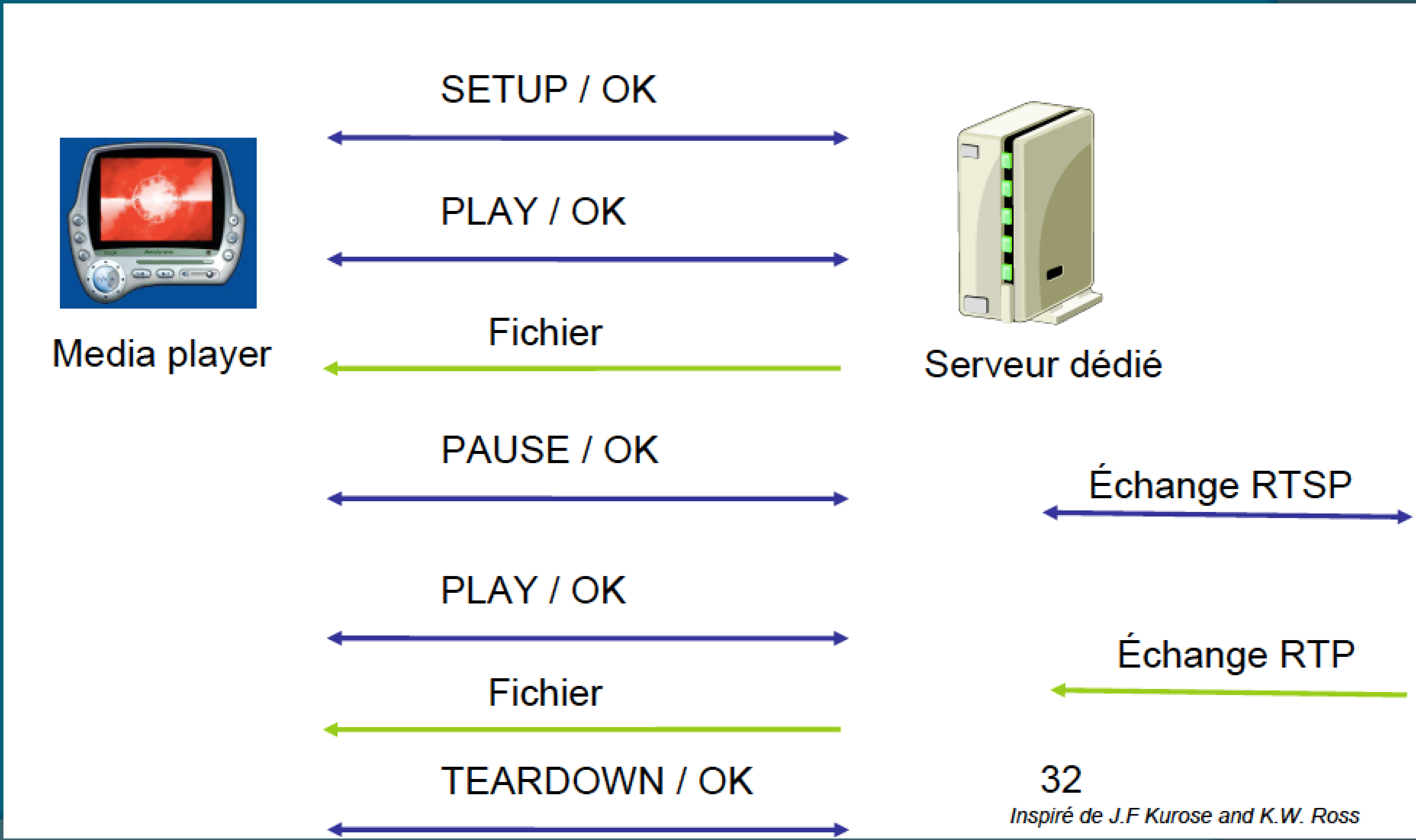
– Choice of transport protocol

– Choice of technique for buffering

• What he does

– Help the mediaplayerto control the transmission of an audio/video stream

Flow send control



RTSP session

Session ID chosen by the server

- Used in every message
- History of the state of the client at the server
- Stateful Protocol
- RTSP over UDP or TCP

SIP
Session Initiation Protocol

- Lightweight signaling protocol
- ModelClient server
-Mechanisms for establishing/terminating a call on an IP network
  - Prevent the called party from the call
  - Agree on encoding
  - End a call

-Mechanisms for determining which called party's IP address to use
- IP address not necessarily fixed
  - Mobility
  - Multi-terminals

- Call management
  - Changing encoding during a call
  - Invite other participants
  - Call transfer, etc.

SIP method

- Specified in the first bytes of SIP requests
- Indicates the purpose of the message
- GUEST
  - To initiate a session
- ACK
  - To confirm session establishment
  - Used with INVITE
- CANCEL
  - To cancel a pending INVITE request
- BYE
  - To end a session
- REGISTER
  - To register