Network & Media Laboratory

Fundamentals of Multimedia

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Lecture 13
Multimedia Network Communication and Application

Content

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  - Quality of Service (QoS)
  - QoS for IP Protocols
  - Prioritized Delivery
- Multimedia over IP
  - IP-Multicast
  - RTP/RTCP; RSVP
  - RTSP; VOIP
- Media-On-Demand，MOD

1. Quality of Multimedia Data Transmission

1.1 Quality of Service (QoS)

Some most important parameters for QoS

- **Data rate**: Mbps
  - A measure of transmission speed kbps
- **Latency**: ms
  - Maximum frame/packet delay
- **PLR**:
  - Packet Loss Rate
- **Delay Jitter**:
  - A measure of smoothness of the audio/video playback
- **Sync Skew**: ms
  - A measure of multimedia data synchronization
  - Usually 20ms~±200ms can be acceptable

1.1 Quality of Service (QoS)

- Delay Jitter

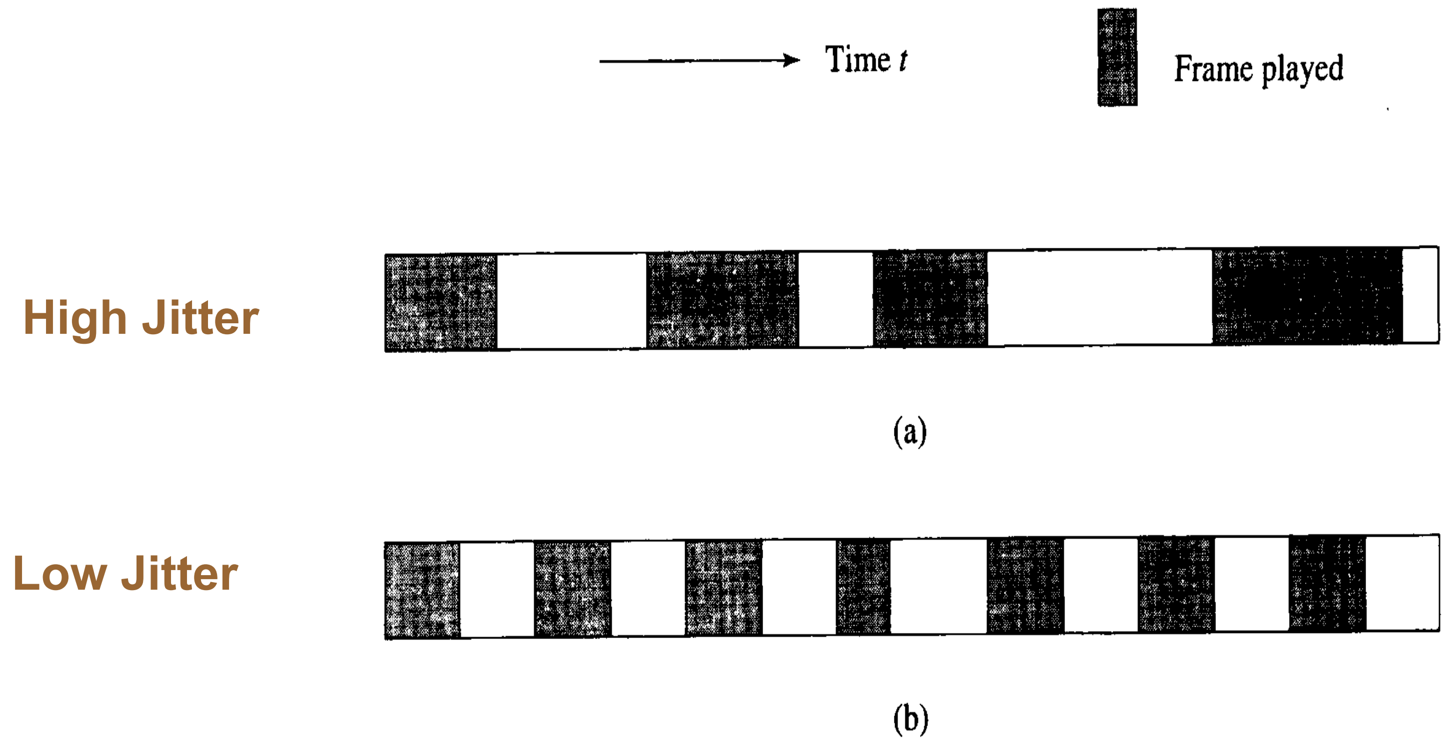


FIGURE 16.1: Jitters in frame playback: (a) high jitter; (b) low jitter.

1.1 Quality of Service (QoS)

Types of multimedia applications:

- **Real-Time/Conversational**: Two-way traffic, low latency and jitter, like voice telephony and video telephony.
- **Priority data**: Two-way traffic, low loss and low latency, like e-commence application
- **Silver**: Moderate latency and jitter, strict ordering and sync. like streaming video, internet games
- **Best-Effort**: No real-time requirement, downloading or transferring large files
- **Bronze**: No guarantees for transmission

1.1 Quality of Service (QoS)

- **Perceived QoS**
  - **Regularity is more important than latency.**
  - **Temporal correctness is more important than the sound and picture quality.**
  - **User focus is usually at the center of the screen, and it takes time to refocus, especially after a scene change.**

Many issues of perception can be exploited in achieving the best perceived QoS in networked Multimedia.

1.2 QoS for IP Protocols

- Status of QoS:
  - Frame relay and ATM provide some levels of QoS.
  - IP is a Best-Effort communication technique, it's hard to provide QoS by current routing methods.

- Could abundant bandwidth provide IP QoS?
  - Abundant bandwidth is not available everywhere.
  - Even if it's available everywhere, bandwidth alone can't resolve problems due to sudden peaks in traffic.

1.2 QoS for IP Protocols

- IETF suggestions for QoS
  - DiffServ: Edge Router, classifies streams according to their applications, core routers transmit data according their priorities.
    - IPv4, uses DiffServ code [TOS（Type of Service）] to enable their differentiated treatment.
    - IPv6: Traffic Class octet to classify the packets

For more details: RFC 2998: http://www.faqs.org/rfcs/

1.2 QoS for IP Protocols

- IETF suggestions for QoS
  - Multiple Protocol Label Switching (MPLS): facilitates the marriage of IP to OSI layer 2 technologies, such as ATM, by overlaying a protocol on top of IP.
  - It inserts one or more shim labels into the header of an IP packet, (Create tunnels, Label Switch Path)
- MPLS: Traffic Engineering
- MPLS: Virtual Private Networks

For more details: RFC 3031 : http://www.faqs.org/rfcs/

1.3 Prioritized Delivery

- **Prioritized Delivery** to alleviate the **perceived deterioration** when high packet loss occur
  - **Prioritization for types of media**： e.g., giving higher priority to audio than to video
  - **Prioritization for uncompressed audio**： sent k of total n groups PCM samples, receiver interpolate the lost groups
  - **Prioritization for JPEG image:** e.g., highest priority for the scan with the DC and first few AC coefficients
  - **Prioritization for compressed video**： e.g., giving the highest priority to I-frames and the lowest priority to B-frames

2. Multimedia Over IP

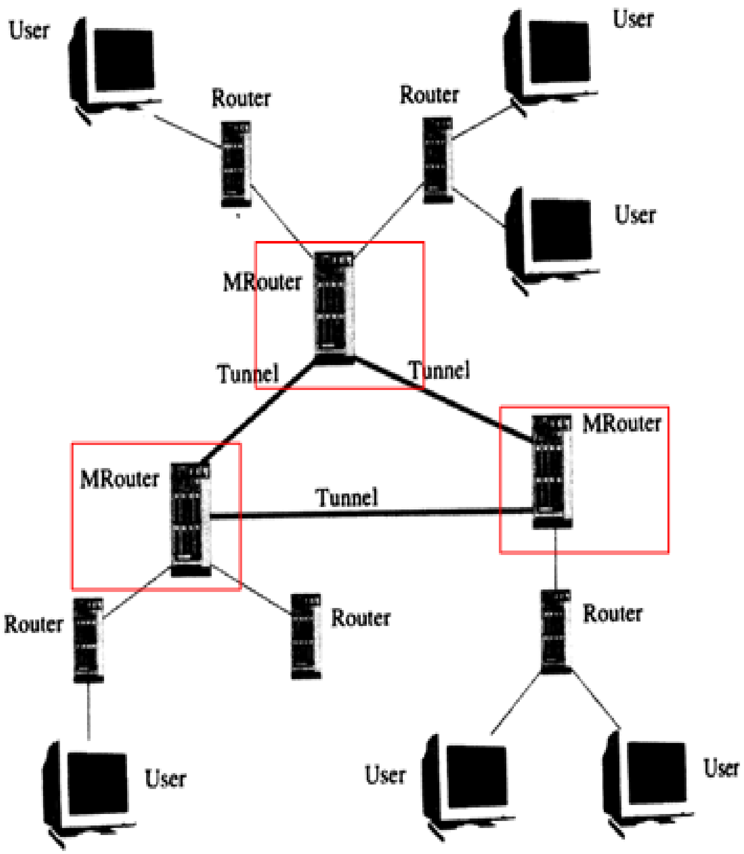
2.1 IP-Multicast

- **Broadcast:** Sent message to all nodes in the domain.
- Unicast: Sent to only one node.
- Multicast: Sent message to a set of specified nodes.

- IP-multicast enables multicast on the Internet
  - Bulletin Boards
  - Group file transfer
  - Audio/video-on-demand
  - ……

2.1 IP-Multicast

- **MBone（Internet Multicast Backbone）**
  - Based on IP-multicast technology
  - Starting in the early 1990
  - Many routers don't support multicast, Mbone uses MRouter to support multicast.



2.1 IP-Multicast

- IPv4 multicast address
  - From 224.0.0.0 ~ 239.255.255.255
  - The host maps IP group addresses into a list of recipients, then multicasts.
    (FDDI and Ethernet have hardware multicast)
- TTL (time to live), if TTL is 0, the packet is discarded
  - TTL to avoid too many packets alive in the network.
- IP-multicast is based on UDP, packets are delivered by "Best-Effort", so reliability is limited.

2.1 IP-Multicast

- IGMP (Internet Group Management Protocol)
  - IGMP was designed to help the maintenance of multicast groups.
  - IGMP has two special types of messages: Query and Report
    - Query messages are multicast by routers to all local hosts, to inquire about group membership.
    - On receiving a query, members wait for a random time before responding.

2.1 IP-Multicast

- Reliable Multicast Transport
  - MBone maintains as a flat virtual topology and does not provide good route aggregation.
  - No central management, ineffective tunnel management.
- RMTP&RMTP II
  - Support forward error control (FEC) , targeted for real-time delivery of multimedia data

2.2 Real-time Transport Protocol

- TCP: Not suitable for real-time multimedia applications
  - Retransmission: introduce great latency and congestion etc
- RTP: For the transport of real-time data, such as (Audio/Video) streams
  - RTP can be applied to multicast or unicast
  - RTP runs on top of UDP
  - UDP not guarantee data packet arrive in original order
  - RTP create its own timestamping and sequencing mechanisms to ensure the ordering

2.2 Real-time Transport Protocol

- RTP Head

00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
Ver		P	X	CC				M	PT							Sequence Number															
Timestamp																															
SSRC																															
CSRC [0..15] :::																															

  - **Bit0 and 1: version of RTP**
  - **Bit2: signals a padded payload**
  - **Bit3: signals an extension to RTP Header**
  - **Bit4 through 7: CSRC count indicates the number of CSRC IDs following the fixed part of the header**
  - **Bit8: signals the first packet in an audio frame or last packet in a video frame**

2.2 Real-time Transport Protocol

- **Bit 9 through 15 (PT) Payload Type (7 bits):**
  - **Indicates the media data type as well as its encoding scheme, so that the receiver knows how to decode it.**

  - Payload type 0: PCM mu-law, 64 kbps
  - Payload type 3, GSM, 13 kbps
  - Payload type 7, LPC, 2.4 kbps
  - Payload type 26, Motion JPEG
  - Payload type 31. H.261
  - Payload type 33, MPEG2 video

PT values of different encoding types

2.2 Real-time Transport Protocol

- **Bit16 through 31 Sequence Number (16 bits):**
  - **Incremented by 1 for each RTP data packet sent**
  - **Check if the data packets arrive in the original order or some packets are lost.**

2.2 Real-time Transport Protocol

- Timestamp field (32 bytes):
  - The most important mechanism of RTP
  - The sender use timestamp records the instant when the first octet of the packet is sampled
  - The receiver play the audio/video in proper timing order and synchronize multiple streams

2.2 Real-time Transport Protocol

- **SSRC field (32 bits):**
  - Identifies sources of multimedia data. Each RTP stream in a conversation has different SSRC ID
- **CSRC field (variable length)**
  - Identifies the source of contributors, such as all speakers in an audio conference

2.3 Real-Time Control Protocol RTCP

- **RTCP: A companion protocol of RTP**

  (The same IP, Different Port)

  - **RTP itself don't provide QoS guarantee.**
  - **RTCP provide information of QoS feedback**
  - **The sender can adjust the strategy of transmission**

- **5 types of RCTP packets:**

00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
Version		P	Count					Type								Length															
Data :::																															

2.3 Real-Time Control Protocol RTCP

- **Receiver Report （RR）：**
  - Provides quality feedback (number of lost packets, etc.)
- **Sender Report （SR）：**
  - Provides information about the reception of RR, number of packets/bytes sent, and so on.
- **Source description （SDES）：**
  - Information about the source, such as e-mail, phone number, name of the participant

2.3 Real-Time Control Protocol RTCP

- **Bye:**
  - **The end of participation.**
- **Application specific functions（APP）：**
  - **Provides for future extension of new features**

2.4 RSVP

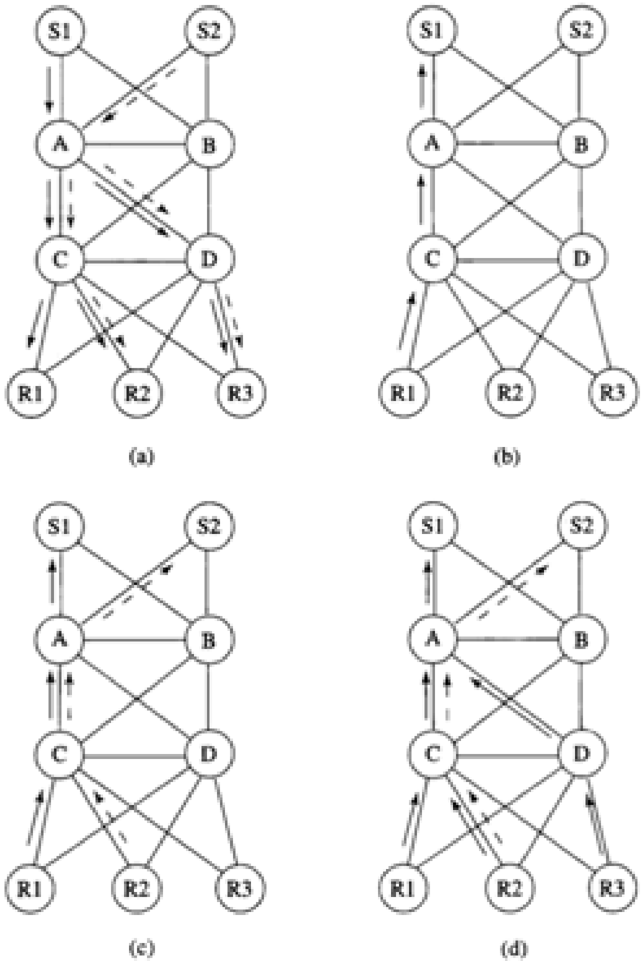
- **RSVP (Resource Reservation protocol)**
  - RSVP developed to guarantee desirable QoS
  - Mostly for multicast, also applicable to unicast
  - Main challenges of RSVP
    - Senders and receivers compete for the limited network bandwidth
    - Receivers can be heterogeneous in demanding different contents with different QoS
    - They can be dynamic joining or quitting multicast groups.

2.4 RSVP

- The most important messages of RSVP:
  - Path: Initiated by the sender and travels towards the multicast destination addresses. Contains information about the sender and the path.
  - Resv: sent by receiver that wishes to make a reservation.
- RSVP receiver-initiated
- RSVP creates only soft state
  - The receiver host must maintain the soft state by periodically sending the same Resv message

2.4 RSVP

- e.g. m Senders and n Receivers in various multicast groups
  - (a) S1 and S2 send PATH message to all receivers
  - (b)R1 sends RESV message to S1
  - (c)R2 sends RESV message to S2
  - (d)R2 and R3 send RESV messages to S1



2.5 Real-Time Streaming Protocol (RTSP)

- Traditionary, Play-after-Downloading
  - Download the whole file, then playback
- Play-While-Downloading
  - The buffer is filled to a certain extent, uncompress the buffer data then playback
  - The buffer space needs to be sufficiently large to deal with the possible jitter and to produce continuous, smooth playback

- RTSP (Real-Time Streaming Protocol) is for communication between a client and a stored media sever

2.5 Real-Time Streaming Protocol (RTSP)

- Four RTSP operations:
  - Requesting presentation description
  - Session setup
  - Requesting and receiving media
  - Session closure

2.6 Internet Telephony (VOIP)

- Public switched telephone network (PSTN) carrying analog voice signals.
  - Provides reliable and low-cost voice and facsimile services.

- PCs and Internet became readily available and more voice data become digital
  - VOIP attract a great deal of interest in research and communications.

2.6 Internet Telephony (VOIP)

- Main advantages of VOIP:
  - Provides great flexibility and extensibility in accommodating integrated services: Voicemail, audio/video conferences.
  - Packet switching, not circuit switching; network usage is much more efficient
  - Technologies of multicast or multipoint communication.
  - Support various degrees of QoS
  - Good graphics user interfaces

2.6 Internet Telephony (VOIP)

- Internet telephony is not simply a streaming media service over the internet, it requires a sophisticated signaling protocol:

H.323 or SIP
RTP, RTCP, RSVP, RTSP
Transport Layer (UDP, TCP)
Network Layer (IP, IP Multicast)
Data Link Layer
Physical Layer

2.6 Internet Telephony (VOIP)

- **H.323**: A standard for packet-based multimedia communication services over networks, don't provide a guaranteed QoS.
  - **Call Setup**: The caller sends the (GateKeeper, GK) a request message, the GK may either grant permission or reject the request.
  - **Capability Exchange**: An H.245 control channel is established to exchange capabilities of both the caller and callee.

2.6 Internet Telephony (VOIP)

- H.323 signaling and control
  - H.255：Call control protocol, including signaling, registration, admissions, packetization and so on
  - H.245：Control protocol for multimedia communications. (opening and closing channels for media streams…)
  - H.235：Security and encryption for H.323

2.6 Internet Telephony (VOIP)

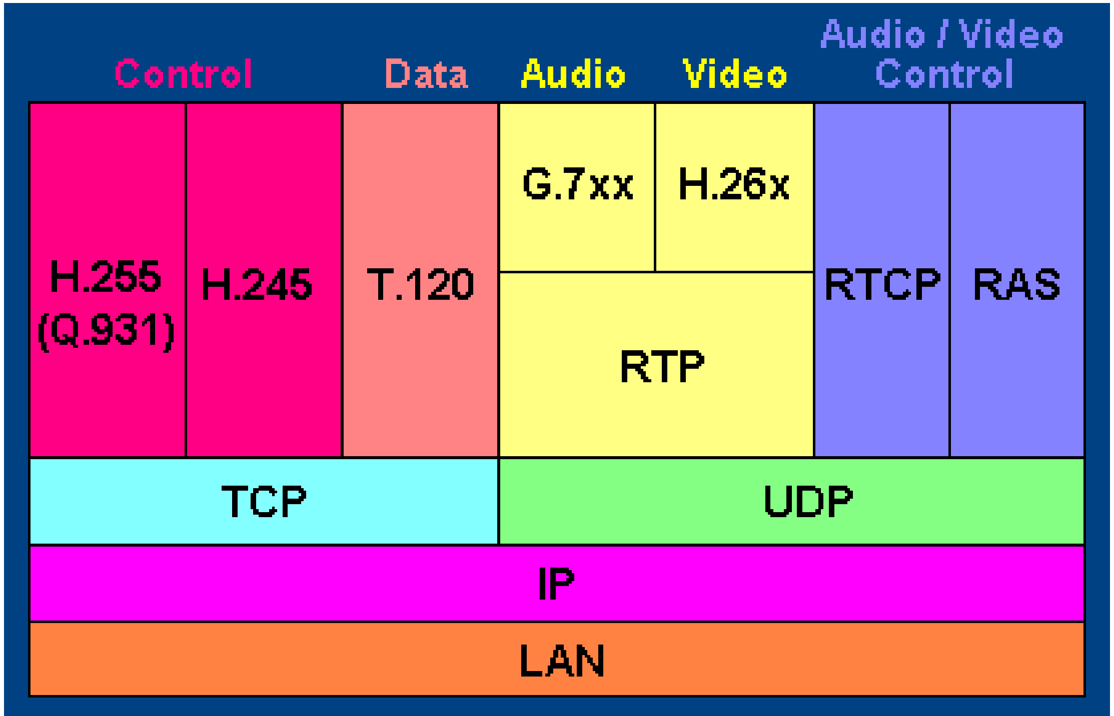
- Audio Codecs:
  - G.711: Codec for 3.1kHz audio over 48,56,64 kbps channels.
  - G.722: Codec for 7kHz audio over 48,56,64 kbps channels.
  - G.723.1: Codec for 3.1kHz, audio over 5.3,6.3 kbps channels.
  - G.728: Codec for 3.1kHz, audio over 16kbps channels.
  - G.729, G.729a: Codec for 3.1kHz, audio over 8kbps channels.

2.6 Internet Telephony (VOIP)

- Video Codecs：
  - H.261： Codec for video at p*64 kbps（$p \geq 1$）
  - H.263： Codec for low-bitrate video (＜64kpbs)
- Related Standards
  - H.320: The original standard for videoconferencing over ISDN networks
  - H.324: An extension of H.320 for video conferencing over the GSTN
  - T.120: Real-time data and conferencing control

2.6 Internet Telephony (VOIP)

- **H.323 Protocol Stack:**



H.323 Protocol Stack

2.6 Internet Telephony (VOIP)

- Session Initiation Protocol (SIP)
  - SIP is an application layer control protocol in charge of establishing and terminating sessions in Internet telephony.
  - SIP is not limited to VOIP, it's also used for multimedia conferences and multimedia distribution.
  - Similar to HTTP, SIP is a text-based protocol that is different from H.323.
  - 3 types of SIP severs:
    - Proxy Server
    - Redirect Server
    - Location Server

2.6 Internet Telephony (VOIP)

- The methods for clients to invoke are:
  - INVITE: invites callee(s) to participate in a call
  - ACK: acknowledges the invitation
  - OPTIONS: inquires about media capabilities without setting up a call.
  - CANCEL: terminates the invitation
  - BYE: terminate a call
  - REGISTER: sends user's location information to a registrar (a SIP server)

2.6 Internet Telephony (VOIP)

- **Session Description Protocol (SDP): describes multimedia sessions in textual form.**
  - **Number and types of media streams (audio, video, whiteboard, etc)**
  - **Destination addresses (Uicast or Multicast)**
  - **Sending and receiving port numbers**
  - **Media formats (Payload Types)**
  - **……**

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3. Media-On-Demand

3.1 Interactive TV (ITV) and Set-Top Box (STB)

- Interactive TV is a multimedia system based on the television set in homes, support a growing number of activities:
  - TV (basic, subscription, pay-per-view)
  - Video-on-Demand (VOD)
  - Information services (news, weather, magazines, sports events, etc)
  - Interactive entertainment (Internet games, etc)
  - E-commerce (online shopping, stock trading)
  - Access to digital libraries and educational materials
  - ……

3.1 Interactive TV (ITV) and Set-Top Box (STB)

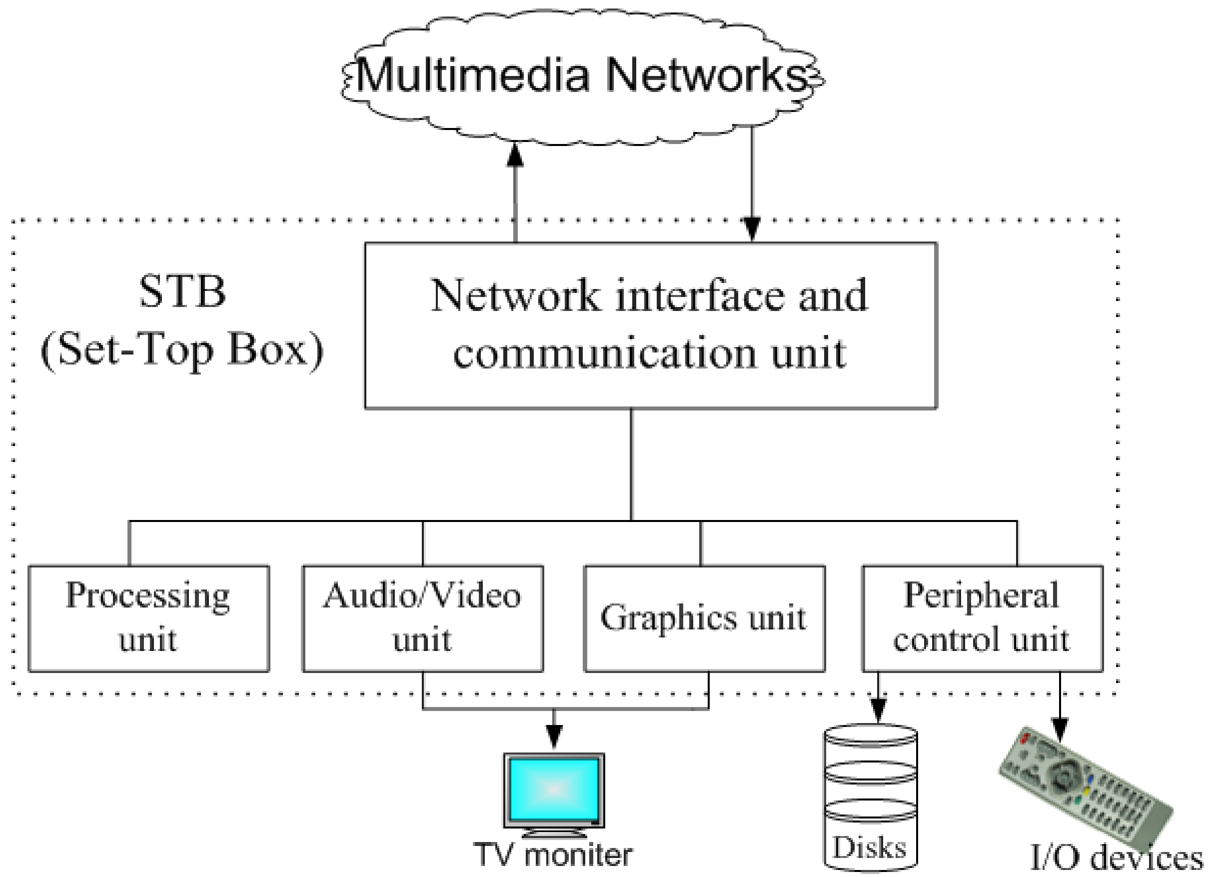
- **Interactive TV techniques:**
  - DVB (Digital Video Broadcasting)
  - Multimedia Home Platform
  - Set-top Box, support bidirectional communications for Interactive TV.

3.1 Interactive TV (ITV) and Set-Top Box (STB)

- **Components of STB :**
  - **Network interface and communication unit**: tuner, demodulator, security devices and so on.
  - **Processing unit**: CPU, memory, operating system.
  - **Audio/video unit**: MPEG-2,4 decoders, DSP, D/A converters.
  - **Graphics unit**: supporting real-time 3D graphics
  - **Peripheral control unit**: controllers for disks, I/O devices, CD/DVD reader and writer and so on.

3.1 Interactive TV (ITV) and Set-Top Box (STB)

- Components of STB



The End!

Thanks!