Standards for multimedia communications

5.1 introduction

• A range of application-level standards have been defined that are concerned with how the integrated information streams associated with the various applications are structured.

• Standards are necessary because it is essential that the two or more items of equipment that are used for the application interpret the integrated information stream in the same way.

• It is necessary also to ensure that both communicating parties utilize the same standards for detecting the presence of bit errors in the received information stream.

• Aspects of communication protocol:
  • Detecting the presence of bit errors in the received information stream and requesting for retransmission.
  • The initiation and clearing of a communications session between 2 communicating applications
  • The setting up and clearing of a connection through the particular network being used.

• In this chapter, we present an overview of the standards that have been defined for use with multimedia communications.

• A common framework known as a reference model is used for defining the various standards.

5.2 reference model

• The standards associated with the 3 types of basic applications have a common structure.

• Functionality of set of standards:
  • Application standards: provide users, through an appropriate interface, with access to a range of multimedia communication applications.
  • Network interface standards: different types of network operate in different modes and each type of network has a different set of standards for interfacing to it.
  • Internal network standards: deal with the internal operation of the network.
In practice, associated with each standard is the set of procedures that are to be used to perform the particular function such as

- How to format the source information stream
- How to detect transmission errors
- How to handle errors

For each function, both communicating parties must adhere to the same set of procedures and collectively these form the *communications protocol* relating to that function.

The implementation of a communication system is based on a layered architecture.

The protocol layers that normally used are based on what is called the *TCP/IP reference model*.
5.2.1 TCP/IP reference model

• A reference model is simply a common framework for defining the specific set of protocols to be used with a particular application/network combination.

• The resulting set of protocols are then known as the protocol stack for that application/network combination.

- Physical layer
  - The physical layer is concerned with how the binary information stream associated with an application is transmitted over the access circuit to the network interface.

- Link layer
  - The more usual form of representing the source information stream is in the form of a contiguous stream of blocks with each block containing the integrated media stream associated with the application.
  - The role of the link layer is to indicate the start and end of each block within the source bitstream and, in a packet-switched network, to add error check bits to the information bitstream for error detection and/or error correction purposes.

- Network layer
  - The network layer is concerned with how the source information stream gets from one end system to another across the total network.
  - Examples:
    - Connection-oriented network: how to set up/clear connection, exchange information
    - Connectionless network: how to format a packet
  - There are different network layer protocols for different types of network.
- **Transport layer**
  
  The role of the transport layer is to mask the differences between the service offered by the various network types from the application layer and instead, to provide the application with a network-independent information interchange service.

- **Application layer**
  
  The application layer provides the user, through a suitable interface, with access to a range of multimedia communication services.

  The application layer in an end system contains a selection of application protocols, each providing a particular service.

5.2.2. Protocol basics

- As the information to be transferred is passed down from one layer to the next, the protocol at each layer adds its own protocol control information (PCI) at the head of what it receives to form a protocol data unit (PDU).

- In the remote system, each layer protocol reads and removes its own PCI from the head and, after interpreting this according to the defined protocol for that layer, passes the remaining information up to the protocol layer immediately above it.
5.3 Standards relating to interpersonal communications

- Interpersonal communications such as telephony, video telephony, data conferencing, and videoconferencing can be provided both by circuit-mode networks and packet-mode networks.

- Most of the standards relating to these applications have been defined by the ITU-T and there are separate standards for different types of network.

5.3.1 Circuit-mode networks

- The network interface standards relate primarily to the physical connection to the network termination and with the procedures followed to set up and clear a connection.

- The basic transport layer function is provided by the multiplexer/demultiplexer.

- The multiplexer merges (1) the source information from the 3 application streams - audio, video and user data - and (2) the system control application into a single stream for transmission over the constant bit rate channel provided by the connection.

*Figure 5.4 Structure of interpersonal communication standards for circuit-mode networks.*
• The system control application is concerned with negotiating and agreeing on the operational parameters to be used with the call/session.

• In a multiparty conference call, it involves each end system communicating with a multipoint control unit (MCU).

• The audio and video codecs each use a particular compression algorithm which is appropriate for the application and within the bandwidth limits provided by the network.

• If the user data is shared between the various members of a conference, the application uses the services provided by a protocol known as a multipoint communications service (MCS).

• A system-level standard embraces a number of additional standards for the various component functions such as audio and video compression.

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* Mandatory

**H.320**

• The H.320 standard is intended for use in end systems that support a range of multimedia applications over an ISDN.
Audio
- Options: G.711 (64kbps), G.722 (64kbps) and G.728 (16kbps)
- Determined primarily by the amount of available bandwidth.

Video
- H.261
- video resolution: QCIF or CIF, negotiable
- a constant bit rate is maintained by varying the quantization threshold.

User data
- Based on T.120 standard
- Application-specific recommendations that support the sharing of various media types
  - T.124: text
  - T.126: still-image and whiteboard
  - T.127: file contents (text and binary)
  - T.128: text documents and spreadsheets

Communications-related recommendations
- T.122: multipoint control unit (MCU) procedures
- T.125: multipoint communication services (MCS) procedures
- To use non-standard protocols is negotiable.

System control/call setup
- The call setup (signaling) procedure associated with an ISDN is defined in Recommendation Q.931.
- It involves the exchange of messages over a separate 16kbps channel known as the signaling channel.
- The bandwidth associated with the audio, video and data streams are negotiated and fixed at the start of a conference.
- Recommendation H.242 is concerned primarily with the negotiation of the bandwidth/bit rate to be used for each stream.
- Once an end system has set up a connection to the MCU, it informs the MCU of its capabilities. The MCU then negotiates and agrees a minimum set of capabilities so that all members of the conference can participate.
Multiplexing

- It is defined in Recommendation H.221 and describes how the audio, video, and data streams are multiplexed together for transmission over the network.
- Time division multiplexing (TDM) technique is normally used.

**H.324**

- The H.324 standard is intended for use in a PSTN.

**Video**

- Options: H.261 or H.263

**Audio**

- Options: either G.723.1 (5.3/6.3kbps) or G.729

**User data:**

- Basically the same set of protocols as are used in an H.320-compliant terminal except for the network-specific transport protocol T.123.

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**Multiplexing**

- Streams are not allocated fixed portions of the available bandwidth but rather these are negotiated using the H.245 system control protocol.
- The total channel bandwidth is divided into a number of separate logical channels each of which is identified by means of a *logical channel number* (LCN).
- The allocation of LCNs is controlled by the transmitter.
- A *bit-oriented protocol* is used to merge streams that are currently present into the available channel.

**Figure 5.5 H.223 multiplex principles:** (a) structure of the multiplexed byte stream; (b) multiplex table usage.
M-PDU is sandwiched by 01111110.

M-PDU is a field containing a header and information.

A field contains a number of logical channels.

Each channel carries separate media stream or control information.

H contains a 4-bit multiplex code which is used to specify a particular pattern of logical channels in the M-PDU.

The multiplex table can be modified by the transmitter if necessary.

**Adaptation**

- Additional bytes can be added by the transmitter for error detection/correction purposes.

- The adaptation layer specified in H.223 standard supports 3 different schemes:
  - AL1: support retransmission, for user data applications
  - AL2: retransmission is optional, for audio and video streams
  - AL3: support retransmission, for serious video applications

- All schemes support error detection.

**Multipoint conferencing**

- The H.324 standard supports multipoint conferencing via an MCU.

- MCU negotiates an agreed minimum bit rate with all the participants by the exchange of system control messages.

- Internetworking between an H.324 terminal and an H.320 terminal can be supported.

- In such a case, transcoding for audio stream may be necessary.
System control

- The H.245 system control standard is concerned with the overall control of the end system.
- Functions involved:
  - Exchange of messages for the negotiation of capabilities
  - Opening and closing of logical channels
  - Transmission of the contents of the multiplex table
  - Choice of adaptation layers

5.3.2 Packet-switched networks

- Two alternative sets of protocols have been defined for providing interpersonal communication services over packet-switched networks:
  - ITU recommendation H.323
  - IETF standards

**H.323**

- Unlike the H.322 standard which relates to LANS that offer a guaranteed bandwidth/QoS, the H.323 standard is intended for use with LANs that provide a non-guaranteed QoS which, in practice, applies to the majority of LANs.
- The standard comprises components for the packetization and synchronization of the audio and video streams, an admission control procedure for end systems to join a conference, multipoint conference control, and interworking with terminals that are connected to the different types of circuit-switched networks.
- The standard is independent of the underlying transport and network interface protocols and hence can be used with any type of LAN.
Audio and video coding

- **Audio**:
  - Options: G.711 or G.728 when work with H.320-compliant terminals
  - G.723.1 or G.729 when work with H.324-compliant terminals.

- **Video**:
  - Options: either H.261 or H.263, negotiated prior to transmission

- The compressed audio and video streams are formatted into packets for transfer over the network using the *real-time transport protocol* (RTP).

- RTP is for the transfer of real-time information.

- At the head of each RTP packet, there is a format specification which defines how the packet contents are structured.
• A sending end system does the following with the Real-time transport control protocol (RTCP).

• Send information to allow the receiving end system to synchronize the audio and video streams

• Send information such as the transmitted packet rate, the packet transmission delay, the percentage of packets lost/corrupted and the interarrival jitter such that the corresponding end systems can use them to optimize the number and size of receiver buffers and to determine if the retransmission of lost packets is feasible.

Call setup

• LANs do not provide a guaranteed QoS and have no procedures to limit the number of calls/sessions that are using the LAN concurrently.

• In order to limit the number of concurrent calls that involve multimedia, a device called an H.323 gatekeeper can (optionally) be used.

• To set up a call or request additional bandwidth, each end system must first obtain permission from the gatekeeper.

• The messages exchanged with the gatekeeper concerned with the 2 end systems obtaining permission to set up a call are part of the resource access service (RAS) protocol.

**Figure 5.7 Two-party call setup procedure using an H.323 gatekeeper.**

<table>
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<tr>
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<td></td>
<td></td>
</tr>
</tbody>
</table>

resource access service (RAS) messages

Q.931 signaling messages

ARQ = access request
ACF = access confirm
ARJ = access reject
Interworking

- The H.323 standard also defines how internetworking with end systems that are attached to a circuit-mode network is achieved through a H.323 gateway.

- The role of a gateway is to provide translations between the different procedures associated with each network type.

- In order to minimize the amount of transcoding required in the gateway, the same audio and video codec standards are used whenever possible.

- A second function associated with a gateway relates to address translation.

  - Different types of network use different addressing schemes. (e.g. IP address in a LAN using the TCP/IP protocol set and telephone number in a PSTN.)

  - The gatekeeper performs the necessary translation between the different address types during the call setup procedure.
Summary of standards identified/discussed