MULTIMEDIA NETWORK PROTOCOLS

Ing. Omer SALIH, Doctoral Degree Programme (2)
Dept. of Telecommunications, FEEC, BUT
E-mail: saliho@feec.vutbr.cz

Supervised by: Dr. Vít Novotný

ABSTRACT

In this paper we will try to describe the Network of Multimedia as the Voice, Data and Video, all integrated together under the new name MULTIMEDIA and the Network structure such as Protocols. Voice Quality is still suffering for transfer easily and with good quality. The Internet services are done under networks connection services only. So we should treat the problems of networks to optimize the Multimedia services.

1 INTRODUCTION

Multimedia is the combined use of several media, such as movies, slides, music, and lighting, especially for the purpose of education or entertainment. Networked multimedia is to build the multimedia on network and distributed systems, so different users on different machines can share image, sound, video, voice and many other features and to communicate with each under these tools. The use of IP for voice and multimedia applications such as data and video is growing at a rapid pace as the business incentives become increasingly attractive to consumers, businesses and Service Providers.

2 VOIP (VOICE OVER IP)

With Internet Protocol (IP) multicast addressing at the network layer the service group communication can be established across the Internet.

VoIP is simply the transport of voice traffic by using the Internet Protocol (IP), and this definition is hardly surprising.

High-availability solutions for VoIP networks address the need for users to be able to place and receive calls under peak-load call rates or during device maintenance or failure. In addition to lost productivity, voice-network downtime often results in lost revenue, customer dissatisfaction, and even a weakened market position. Various situations can take devices off line, ranging from planned downtime for maintenance to catastrophic failure.

There are two key elements that contribute to availability in a VoIP network: capacity and redundancy. These concepts will now be explored further.

Capacity is a measurement of the volume of traffic a network is engineered to handle. Voice networks are typically engineered to handle a target peak-load capacity, commonly measured in calls per second. Target peak–load capacities are specific to each business and industry, and are based on measured busy–hour call rates. For example, the traffic during the busiest hour on Mother's Day may be the target peak-load capacity for a residential voice network.

Redundancy measures the extra capacity, to be used only in the event of an equipment failure that is placed in a network. When a primary node in a voice network is taken down for maintenance or failure, a redundant secondary device can take over the processing of the voice traffic.

2.1 VOICE QUALITY

Networked multimedia face many technical challenges like real-time data over non-real time network, high data rate over limited network bandwidth, unpredictable availability of network bandwidth. High-bandwidth network protocols such as 100–Mbps Ethernet, FDDI, and ATM are expected to make the networking of digital video and audio practical. The basis of Internet, TCP/IP, provides the range of services needed to support both small and large-scale networks.

There are some keys to acceptable voice quality such as bandwidth. We would like to minimize bandwidth while maintaining sufficiently good voice quality. Bandwidth is easily quantified, but how do we quantify voice quality? Surely, this measurement is subjective rather than objective. Standardized methods exist for measuring voice quality, however, including a standardized ranking system called the Mean Opinion Score (MOS) by ITU. Basically, MOS is a five-point scale as follows:

Excellent	**5**
Good	**4**
Fair	**3**
Poor	**2**
Bad	**1**

Tab. 1: *MOS is a five-point scale*

The objective with any coding technique is to achieve as high a ranking on this scale as possible while keeping the bandwidth requirement relatively low.

In general, and within design restrictions such as bandwidth, most coding schemes aim to achieve or approach toll quality.

The quality of voice is greatly affected by latency and jitter and a packet network. Therefore, it is important for network designers to consider implementation of QoS polices on the network. In addition to protecting voice from data, this has the added benefit of protecting critical data applications from bandwidth starvation because of over subscription of voice calls.

The elements of good QoS design include provisions for managing packet loss, delay, jitter, and bandwidth efficiency. Tools used to accomplish these goals are defined here:

- **Policing**: Provides simple limiting of packet rate, often by simply dropping packets that exceed thresholds to match capacities between different network elements.
- **Traffic shaping:** Provides the capability to buffer and smooth traffic flows into and out of devices based on packet rate.
- **Call admission:** Provides the capability to reject requests for network bandwidth from application, an example might be the use of RSVP.
- **Queuing/scheduling:** There are used with buffering to determine the priority of packets to be transmitted.
- **Tagging/marking:** Includes various techniques to identify packets for special handling
- **Fragmentation:** Refers to the capability of some network devices to subdivide large packets into smaller ones before traversing a narrow bandwidth link.

3 MULTIMEDIA PROTOCOLS

The use of TCP and UDP on IP to transport data is not an effective method of supporting multimedia applications. UDP provides no ordering or reliability and cannot support the streaming nature of the media. TCP can provide ordering and reliability, however it cannot bind jitter and also provides overhead in large packet headers. Figure-1 shows the Multimedia Protocols.

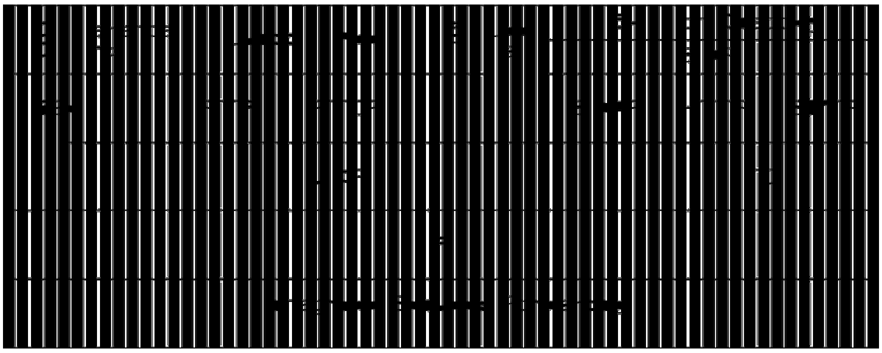

Fig. 1: *Multimedia Protocols*

The Internet has been used primarily for the reliable transmission of data with minimal or no delay constraints. The TCP/IP protocols were designed for this type of traffic and work very well in this context. In addition, the "slow start" TCP congestion-control mechanism can interfere with the audio and video "natural" playout rate. Since there is no fixed path for datagrams to flow across the Internet, there is no mechanism for ensuring that the bandwidth needed for multimedia is available between the sender and receiver(s), so QoS cannot be guaranteed. Introduced below are the RTP, RSVP and RTSP protocols. Since many real-time applications can conserve network and server resources by using IP Multicast, the special requirements and characteristics of IP Multicast have been considered in the design of these protocols, such as scalability, multicast routing, and accommodation of large numbers of receivers and heterogeneous receivers. The issue to overcome is to create protocols that can provide lightweight communication that can deal with the transportation of multimedia data.

3.1 TRANSPORT PROTOCOLS FOR MULTIMEDIA

The transport of real–time multimedia, to large audiences, implies a number of requirements:

- Separate Flows, (multicast groups), for each media stream.
- Receiver adaptation. Applications must be given ability to even out Jitter.
- Synchronization. The relationship in time between media must be maintained.
- The Real–time Transport Protocol – RTP – may be provides this.

3.1.1 REAL-TIME STREAM PROTOCOL (RTSP)

It is a client-server multimedia presentation control protocol, designed to address the needs for efficient delivery of streamed multimedia over IP networks.

3.1.2 RESOURCE RESERVATION PROTOCOL (RSVP)

It is a resource reservation setup protocol designed for an integrated services internetwork.

3.1.3 REAL TIME PROTOCOL (RTP).

It is a real–time transport protocol that provides end–to–end delivery services to support applications transmitting real-time data.

Requirements like end-to-end delay, delay jitter, synchronization, and reliability cannot be directly met by network services; instead, end–to–end protocols are necessary to provide the requested QoS. Existing end-to-end protocols - located in layers 4, 5, 6, and 7 of the OSI Reference Model – represent a bottleneck in the communication process and cannot provide the performance of high-speed networks unaltered to the applications. One way of dealing with these problems is in the use of dynamic protocol stacks. Where a framework for dynamic protocol configuration and adaptation is constructed. Constructing the appropriate protocol stack for each connection allows the minimal stack, which is best, suited to the environment. Communication subsystems like those described in many research projects, try to solve the problems by introducing highly flexible communication subsystems and a protocol configuration approach. Complex protocols are decomposed into fine granular building blocks each defining a single protocol task. The goal of the configuration is to combine building blocks in such a manner that the resulting protocol configuration is as light as possible and fulfills the application requirements.

H.323 is an umbrella recommendation from the International Telecommunications Union (ITU) that sets standards for multimedia communications over Local Area Networks (LANs) that do not provide a guaranteed Quality of Service (QoS).

4 PACKET-SWITCHED NETWORKS

With packet multimedia data there is no need for the different media comprising a session to be carried in the same packets. In fact it simplifies receivers if different media streams are carried in separate flows (i.e., separate transport ports and/or separate multicast groups). This also allows the different media to be given different quality of service QoS. For example, under congestion, a router might preferentially drop video packets over audio packets. In addition, some sites may not wish to receive all the media flows.

UMTS is a so-called "third-generation (3G)," broadband, packet-based transmission of text, digitized voice, video, and multimedia at data rates up to and possibly higher than 2 megabits per second (Mbps), offering a consistent set of services to mobile computer and phone users no matter where they are located in the world.

Compelling services to mobile phones will drive new revenue streams regardless of whether they are rolled out over 2.5G or 3G. In European markets, messaging is currently a major revenue generator for data and this is likely to continue to be the case in the first phase of packet–switched networks.MultiMedia Messaging (MMS) is an extremely good example of a service enabled on packet-switched networks and of the evolutionary nature of emerging technologies. MMS provides the capability of sending rich multimedia content to mobile phones and is generally seen as one of the key potential revenue streams for network operators. MMS is a 3G technology. However, with the delays in 3G, the introduction of MMS has been brought forward to 2.5G networks and is being launched using WAP as a bearer. Initial implementations of MMS will produce messages including video or animated graphics and sounds. Future versions will include more complex formats and contents such as video streaming, which will be possible once MMS moves to an IP-bearer. The complexity and richness of these services drives the need for an advanced, open, standard, mobile operating system.

The IP Multimedia Core combined with terminals supporting Java downloads creates exiting new opportunities for application developers and operators to develop and offer exciting new IP multimedia services.

CONCLUSION

In this paper we described the Multimedia Networks. We concentrated on Voice over IP (VoIP), which appears as the tool of the transmission through the Internet connection. The Internet services are done under networks connection services only. So we described the protocols of multimedia such as VoIP, RTSP, RSVP and RTP to optimize the Multimedia services. With Packet-Switched Network packets multimedia data, there is no need for the different media comprising a session to be carried in the same packets.

REFERENCES

[1] http://searchdatabase.techtarget.com/sDefinition/0,,sid13_gci211899,00.html1

[2] Cisco Systems, et. Al. “Internetworking Technologies Handbook, Third edition” Cisco Press June 2001

[3] Yi-Bing Lin, Imrich Chlarmtac “Wireless and Mobile Network Architectures” Wiley 2001.

[4] DANIEL COLLINS “Carrier Grade Voice Over IP” McGRAW-HILL 2001

[5] http://www.nokia.com/cda1/0,1080,2035,00.html

[6] http://www-mice.cs.ucl.ac.uk/multimedia/projects/mice/mbone_review.html1

[7] http://www.gweep.net/~rocko/mqp/node11.html#401

[8] SIP “http://www.argreenhouse.com/SIP-mobile/sip-hetero-3g.pdf”