

Broadcasting video over the Cellular network and the Internet

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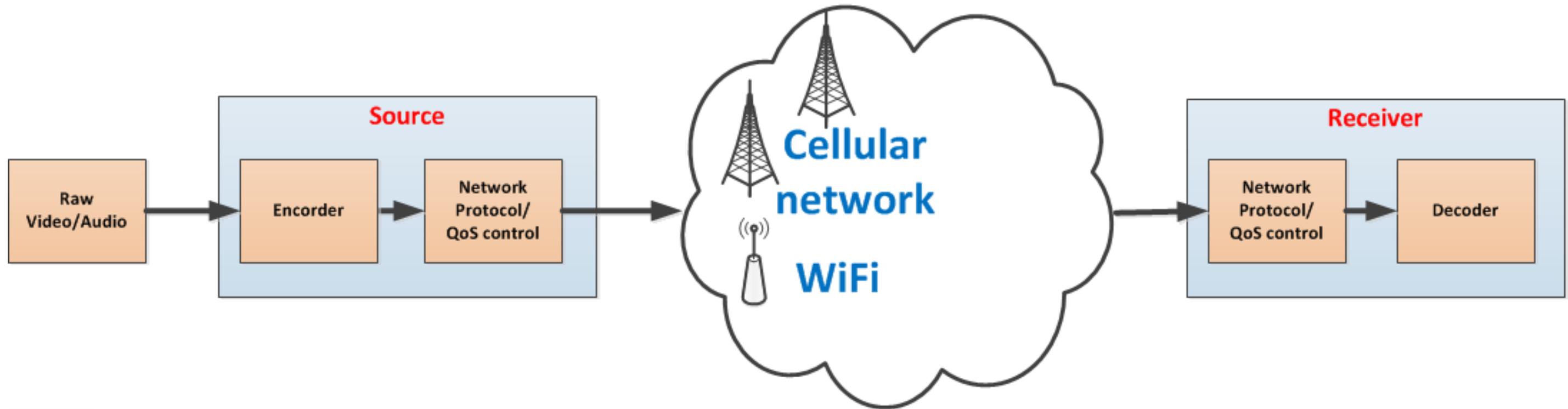
Introduction

- Huge chunks of the off-air broadcast and microwave television spectrum have **slowly and systematically been lopped off**.
- All the while, the volume of news gathering content and distribution has dramatically increased.
- How can the broadcaster function in the midst of this escalating and diametrically opposed broadcast environment?
- The answer is through increased efficiencies in modulation and encoding techniques and leveraged augmentation of **3G/4G wireless infrastructures**.
- However compared to traditional video transmission, the best effort nature of these packetized networks pose many challenges.



Streaming over Cellular Network and Internet

Cellular network and the Internet



Streaming over Cellular Network and Internet

- Raw video must be **processed, graded** and **compressed** before transmission to achieve efficiency.
- **H.264/AVC** encoding standards provide good video quality at substantially lower video bit rates.
- **Adaptive Bit Rate (ABR)** encoding maximizes the video over IP performance by dynamically adjusting the H.264/AVC encoding real time rate control parameters.
- Network protocols such as **Internet Protocol (IP)**, **Transmission Control Protocol (TCP)** and **User Datagram Protocol (UDP)** are used to stream video between the end users with cell phones, tablets or laptops.
- **Congestion control** and **error control** are usually needed to cope with packet loss and delay in wired or wireless IP networks.



Encoding

- To maximize the video performance over IP networks, **H.264/AVC encoding is used with adaptive bit rate (ABR) encoding control.**
- H.264/AVC video encoding provides **good video quality at substantially lower video bit rates** versus some of the previous MPEG standards.
- **This attribute is important for network bandwidth constrained applications, for wired or wireless IP networks, such as video over cellular 3G/4G networks.**
- H.264/AVC delivers video at lower bit rates by employing several techniques which include, improving the accuracy of the motion compensated prediction picture which yields bit rate savings by also taking advantage of redundancy that exists between video frames, the ability to describe the motion with the least amount of bits across a long group of video frames.
- **Adaptive Bit Rate (ABR)** encoding maximizes the video over IP performance throughout the network by dynamically adjusting H.264/AVC encoding real time rate control parameters.

Streaming Network Protocols

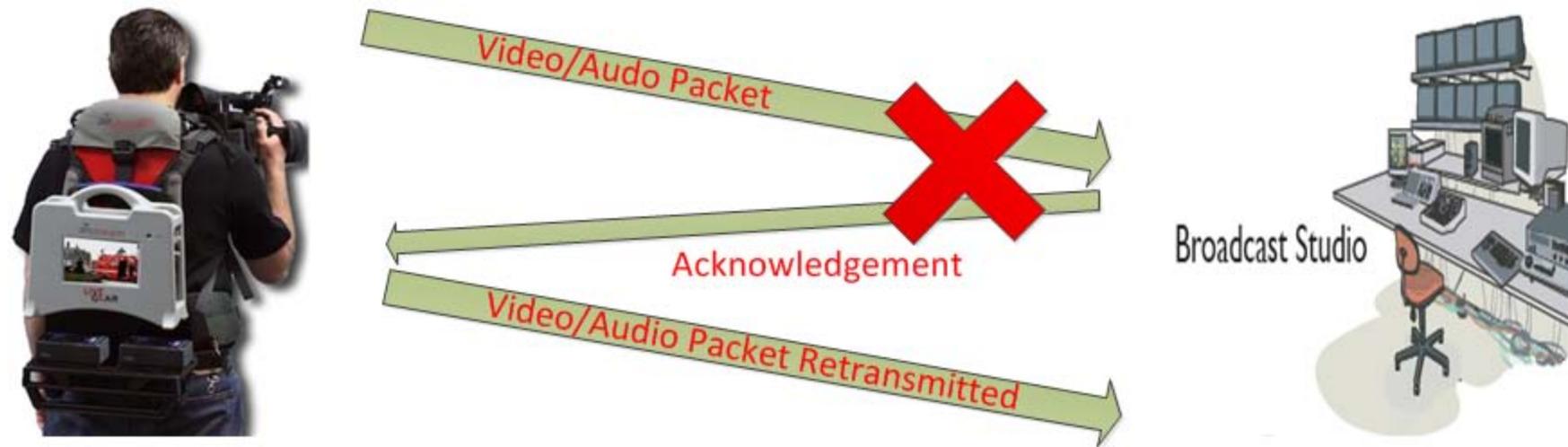
- Various protocols and their combination have been designed and standardized to provide a complete streaming service between the end users over the IP network.
- Protocols used for streaming application can be classified into:
 - **Session Control Protocol:** Protocols such as *Real-Time Streaming Protocol (RTSP)* and *Session Initiation Protocol (SIP)* provide control over delivering video during an established session.
 - **Network Layer Protocol:** Protocol such as the *Internet Protocol (IP)* supports network addressing of the senders and receivers.
 - **Transport Protocol:** provides end-to-end functions for streaming applications. Two fundamental transport protocols used today:
 - *Transmission Control Protocol (TCP)*
 - *User Datagram Protocol (UDP)*

Transmission Control Protocol (TCP)

- TCP provides a **connection-oriented** full – duplex channel between the sender and the receiver.
- It is widely used to transport data over the internet.
- TCP requires **three-way handshaking** to establish a connection between the sender and a receiver.
- After the connection has been established, TCP manages a reliable transmission where the receiver receives the data in the same order the source sent it.
- TCP employs **congestion control** to avoid sending too much traffic and adapt to the load conditions within the network.
- If congestion is experienced, the sender lowers its transfer rate to allow the network to recover.
- TCP also employs **flow control** to prevent the receiver buffer from overflowing.

Transmission Control Protocol (TCP)

- It detects any packet that is **discarded, duplicated, corrupted** or **reordered** by the network.
- The sender retransmits **damaged** or **lost packets** thus allowing the receiver to reconstruct the original data stream.



- TCP tries to achieve the highest possible data-transfer rate without causing significant data loss.

User Datagram Protocol (UDP)

- UDP provides a **connectionless best-effort** service to transfer data within a network. It is unreliable as it provides no guarantee of packet delivery or protection from packet duplications.
- It **does not** provide any **congestion control** and can “choke” the network by sending data at a higher rate than available network capacity.
- However, UDP has minimal overhead as it does not have to establish connections, perform congestion control or error corrections.
- It generally provides better throughput “in network” with reliable physical and data-link layers.
- When adequate, UDP does offer very efficient communication transport to some applications.

Streaming Network Protocols

- Both TCP and UDP have attractive features needed for broadcasting video over the Internet.
- However, TCP might not be efficient as it incurs a lot of overhead, delay and the retransmitted packets might miss its play-out time.
- Whereas, UDP is not reliable and does not have flow control capabilities.
- **Stream Control Transmission Protocol (SCTP)** and **Reliable UDP (RUDP)** attempt to adopt the simple transmission model of UDP, but with the addition of the desirable congestion and error control of TCP.
- Another approach is to use an upper layer protocol such as **Real-Time Transport Protocol (RTP)**, which works on top of UDP/TCP, to provide end-to-end transport functions for streaming applications.

Quality of Service

- **The Internet does not offer the Quality of Service (QoS)** required for streaming real-time video.
- **Excessive delay** and **burst of packet losses** in the Internet can have a major impact on the performance of real-time video streaming applications.
- **Due to its real-time nature, streaming video has bandwidth, delay and timing constraints.**
- In the absence of QoS support from the network, the end users must employ application-layer QoS controls such as:
 - flow control or rate control
 - error control

Rate Control

- Rate control is necessary at the end system to help reduce packet loss and excessive delays.
- It determines the sending rate of the video stream based on available network capacity to avoid congestion and loss of packets.
- Rate controls can be:
 - **Source-based rate control:** source adapts the video transmission rate based upon a feedback from the network. This feedback might be either probe-based or TCP model-based.
 - **Receiver-based rate control:** the receiver regulates the rate of the incoming video stream by adding or dropping channels. The source does not participate in rate control.
 - **Hybrid-based rate control:** sender can adjust the rate on each channel based upon the feedback, as well as the receiver can regulate the incoming rate of the video stream.

Error Control

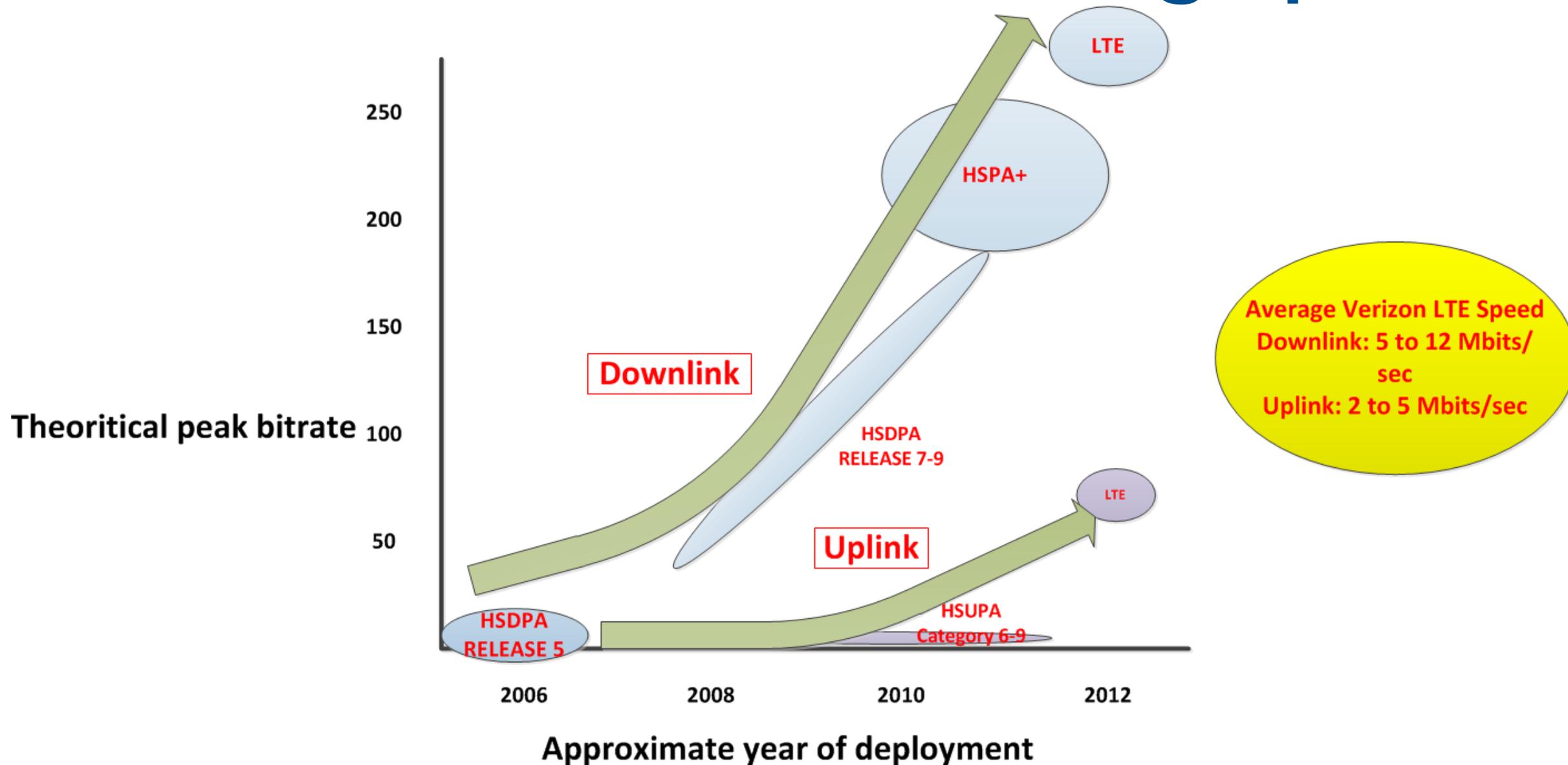
- Error control schemes are necessary for reliable communication as they detect and correct packet errors.
- Error control strategies include:
 - **Forward Error Correction (FEC):** FEC adds additional information on the original message so that it can be constructed correctly at the receiver in the presence of packet loss.
 - **Retransmission:** Retransmission, a key component of reliable protocol, is one of the main methods of recovering lost packets. However, it is not always suitable in real-time video streaming as the retransmission packet may miss its play-out time.
 - **Error concealment:** Its aim is to minimize the deterioration caused by data transmission loss or error by masking the corrupt data.
- For real-time video streaming applications, where latency is a key parameter, FEC schemes are preferable to retransmission-based schemes even though FEC adds redundancy regardless of correct packet receipt or not.

Conclusion

- The cellular/IP network is a **best-effort IP network** which does not offer the Quality of Service (QoS) required live video broadcasting.
- H.264/AVC video encoding can provide good video quality at substantially lower video bit rates. Adaptive Bit Rate (ABR) encoding can further maximize video over IP performance.
- The primary transport protocols, TCP and UDP, both have attractive features, but
- **TCP is inefficient and UDP is unreliable.**
- Other transport protocols (SCTP, RUDP) provide tradeoffs between TCP and UDP.
- **Rate and error control schemes are crucial** to mitigate the loss of packets due to the nature of congested networks which precipitate bursty conditions in wired and wireless packet networks.
- All of these attributes are essential for broadcasting video over bandwidth constrained IP wired and wireless networks , such as 3G/4G cellular networks.



Cellular Network Increasing Speed



Questions

Thank you



Broadcasting video over the Cellular network and the Internet

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Abstract: Because of the ubiquitous nature of the cellular 3G/4G network and the Internet, broadcasting video over the IP networks has gained lot of momentum. However, the lack of Quality of Service (QoS) required to stream real time video makes broadcasting over these networks very challenging.

In bandwidth constraint IP networks, such as the cellular network, video encoding techniques like H.264/AVC provides good video quality at substantially lower video bit rates. Transport protocols have been adopted to provide a complete streaming service between the end users in this packet based network. Excessive delay and bursty packet losses experienced in these networks make QoS control in real-time streaming application crucial. The widely used TCP protocol might not be efficient and its key feature, retransmission of erroneous or lost packets, applicable. UDP protocol, which is generally preferred for real-time applications, is not reliable and lacks flow control capabilities. In this paper, we will focus on the issues faced with broadcasting video stream over the cellular network and the Internet and possible solutions toward achieving low latency broadcast with a certain degree of QoS.

I. Introduction

The ubiquitous nature of the cellular 3G/4G network and the Internet has made broadcasting video over these IP networks very attractive. However compared to traditional video transmission, the best-effort nature of these networks poses many challenges. The Internet does not offer the Quality of Service (QoS) required to streaming real time video.

In streaming application, video is played out while parts of the content are being received and decoded. Due to its real-time nature, streaming video has bandwidth, delay and also timing constraints. The audio and video data must be played out continuously. If there is a pause in streaming because audio or video data did not arrive in time, it is annoying to the end users.

Streaming video over the Internet basically consists of broadcasting content, an encoder to digitize it, protocols to transmit over a content delivery network, such as a cellular network, a wifi hotspot or the Internet, to distribute and deliver the content that meets the Quality of Service (QoS) requirements as shown in figure 1.

Raw video must be processed, graded and compressed before transmission to achieve efficiency. Scalable video coding can achieve high compression efficiency with low complexity. Error resilient encoding can deal with packet loss and delay cognizant video coding is effective in dealing with delay

variations. Combination of these techniques can provide range of solutions to the problem of QoS fluctuations.

Further, H.264/AVC [1] encoding standards provides good video quality at substantially lower video bit rates. Adaptive Bit Rate (ABR) encoding maximizes the video over IP performance by dynamically adjusting the H.264/AVC encoding real time rate control parameters.

Network protocols such as Internal Protocol (IP), Transmission Control Protocol (TCP), and User Datagram Protocol (UDP) are used to streaming video between the end users with cell phones, tablet or laptops. Congestion control and error control are usually needed to cope with packet loss and delay in these wired or wireless IP networks.

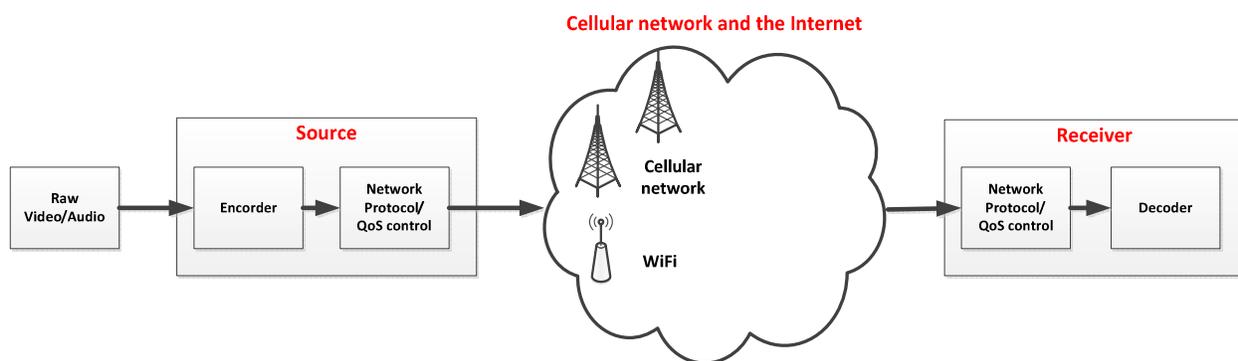


Figure 1: Streaming video of over the cellular network and the Internet

Section II discusses the video encoding techniques. The protocols needed for transporting video stream over the IP network are presented in Section III and Section IV highlights the need of rate and error control. Section V concludes the paper.

II. Adaptive Bit Rate (ABR) Video Encoding Using H.264/AVC

To maximize the video performance over IP networks, H.264/AVC encoding is used with adaptive bit rate (ABR) encoding control.

H.264/AVC video encoding provides good video quality at substantially lower video bit rates versus some of the previous MPEG standards. This attribute is important for network bandwidth constrained applications, for wired or wireless IP networks, such as video over cellular 3G/4G networks. H264/AVC delivers video at lower bit rates by employing several techniques which include, improving the accuracy of the motion compensated prediction pictures which yields bit rate savings by also taking advantage of the redundancy that exists between video frames, the ability to describe the motion with the least amount of bits across a long group of video frames. The third substantial improvement is in the motion compensated prediction at the pixel level. The H.264/AVC standard supports the ability to use quarter pixel precision which doubles the motion prediction accuracy thus allowing for improvements at lower bit rates.

Adaptive bit rate (ABR) encoding maximizes the video over IP performance through the network by dynamically adjusting H.264/AVC encoding real time rate control parameters. A maximum video encoding rate for the network is set as a “cap” value where any encoding rate below this “cap” can be dynamically throttled up and down dependent on the network bandwidth capabilities. A feedback mechanism from the available network bandwidth metric to the H.264/ AVC encoder is applied which allows for real time optimally encoding on the fly by managing the bit allocation within and between the group of pictures (GOP) that is generated within the encoder.

III. Streaming Network Protocol

Various protocols and their combination have been designed and standardized to provide a complete streaming service between the end users over the IP network. Protocols used for streaming application can be mainly classified into session control protocols, network-layer protocols or transport protocols. Session control protocols such as Real-Time Streaming Protocol (RTSP) and Session Initiation Protocol (SIP) provide control over delivering video during an established session. Network layer protocol such as the Internet Protocol (IP) supports network addressing of the sender and receivers.

Transport protocol provides end-to-end functions for streaming applications. Two fundamental transport protocols used today are the Transmission Control Protocol (TCP) and the User Datagram Protocol (UDP):

Transmission Control Protocol (TCP)

TCP provides a connection-oriented full-duplex channel between the sender and the receiver and is widely used to transport data in the Internet. TCP requires a three-way handshaking to establish a connection between a sender and a receiver. The connection is established as follows:

1. The sender sends the remote receiver and initial sequence number using SYN (Synchronize sequence number) packet
2. The remote receiver responds the SYN packet with an Acknowledgment (ACK) packet of the initial sequence number and the initial sequence number of the sender.
3. The sender then responds with an ACK to the receiver’s sequence number.

After the connection has been established, TCP manages a reliable transmission where the receiver receives data in the same order the source sent it. It detects if any packet is discarded by the network, duplicated, corrupted or reordered. The sender maintains a copy of all transmitted packets until it receives an acknowledgement from the receiver of its correct transfer. The sender, then, will retransmit damaged or lost packets thus allowing the receiver to reconstruct the original data stream.

TCP employs congestion control to avoid sending too much traffic and adapt to the load condition within the network. There is no constant data-transfer rate in TCP. If the sender senses that there is more available network capacity, it will increase its transfer rate until some form of network saturation is reached. Conversely, if it experiences congestion, the sender lowers its transfer rate to

allow the network to recover. TCP also employs flow control to prevent the receiver buffer from overflowing. Thus TCP tries to achieve highest possible data-transfer rate without causing significant data loss.

User Datagram Protocol (UDP)

UDP provides a connectionless best-effort service to transfer data within a network. It is unreliable as it provides no guarantees of packet delivery or protections from packet duplications. It does not provide any congestion control and can choke the network by sending data at a higher rate than available network capacity.

However, UDP has minimal overhead as it does not have to establish connections, perform congestion control or error corrections. It generally provides better throughput in network with reliable physical and data-link layers. When adequate, UDP does offer very efficient communication transport to some applications.

Both TCP and UDP have attractive features needed for broadcasting video over the Internet. However, TCP might not be efficient as it incurs a lot of overhead, delay and the retransmitted packets might miss its play-out time whereas UDP is not reliable and does not have flow control capabilities. Transport layer protocols such as *Stream Control Transmission Protocol (SCTP)* [2] and *Reliable UDP (RUDP)* [3] attempt to adopt simple transmission model of UDP but add the desirable congestion control and error control of TCP. Another approach is to use upper layer protocol such as *Real-Time Transport Protocol (RTP)* [4], which works on top of UDP/TCP, to provide end-to-end transport functions for streaming applications.

IV. Quality of Service (QoS) control

Excessive delay and bursty nature of packet losses in the Internet can have a major impact on the performance of real-time video streaming application. In the absence of QoS support from the network, the end users must employ application-layer QoS control such as flow control or rate control and error control.

Rate Control

Rate control is, therefore, necessary at end system to help reduce packet loss and excessive delays. It determines the sending rate of video stream based on available network capacity to avoid congestion and loss of packets. Rate controls can be performed on the source, the receiver or both.

In the *source-based rate control* the sender adapts the video transmission rate based on a feedback about the network. For unicast streaming, this feedback employed by the source can be either probe-based or TCP model-based. In the probe-based approach, the source adjusts its sending rate to match the network capacity by maintaining the packet loss ratio below a certain threshold. The rate can be adjusted using additive increase and multiplicative decrease or multiplicative increase and multiplicative decrease schemes.

The other approach is based on TCP model, where its throughput is calculated using equation 1 based on the round trip time (RTT), maximum transmit unit (MTU) and the packet loss ratio, p , experienced by the connection.

$$\zeta = \frac{1.22 * MTU}{RTT * \sqrt{p}} \quad (1)$$

The source can adjust its video transmission rate with the measured values of RTT and packet loss and avoid congestion similar to TCP.

In the *receiver-based rate control*, the receiver regulates the received rate of the video streams by adding or dropping channels. The source does not participate in rate control. The receiver probes the network to check for congestion. If it does not detect congestion, it adds a layer increasing its receiving rate. It checks for congestion again and if detected, drops the layer resulting on a lower receiving rate. The receiver can also estimate connection through equation 1 and adjust its receiving rate.

In the *hybrid rate control*, the sender can adjust rate on each channel based on the feedback as well as the receiver can regulate its receiving rate for the video stream.

The *receiver-based* and *hybrid rate control* provide good trade-off between bandwidth efficiency and service flexibility for multicast video streaming.

Error Control

Error control schemes are necessary for reliable communication as it detects and corrects packet errors. Error control strategies include forward error correction (FEC), retransmission schemes and error concealment.

Forward Error Correction (FEC) adds additional information on the original message so that they can be constructed correctly at the receiver in the presence of packet loss. Channel coded FEC applies a block code to a set of packets, say k , to generate an n -packet block (where $n > k$). The receiver only needs to recover k packets out of the n -packet block to correctly play out the video stream. Other types of FEC such as source coding-based FEC (SFEC) or joint source/channel coding-based FEC also adds redundant information so that it can be recovered from loss at the receiver.

Retransmission is one of the main methods of recovering lost packets. However, it is not always suitable in real-time video streaming as the retransmitted packet might miss its play-out time. However if the one-way trip time is short with respect to maximum allowable delay, then a delay-constrained retransmission scheme can be used for error control. Retransmission, one of the key components of reliable protocol, is not a viable option if the retransmitted packet does not arrive in time for display.

Error concealment aims to minimize the deterioration caused by data transmission loss or error by masking the corrupt data.

Thus for real-time video streaming applications, where latency is a key parameter, FEC schemes are preferable to retransmission-based schemes even though FEC adds redundancy regardless of correct packet receipt or not.

V. Conclusion

The Internet and the cellular network are a best-effort IP network which does not offer the Quality of Service (QoS) required for broadcasting live video. H.264/AVC video encoding can provide good video quality at substantially lower video bit rates. Adaptive Bit Rate (ABR) encoding can further maximize the video over IP performance through the network by dynamically adjusting H.264/AVC encoding real time rate control parameters. The primary transport protocols TCP and UDP both have attractive features for streaming real-time video over the IP networks. However, the widely-used protocols TCP is not efficient and UDP not reliable.

Other transport protocols such as SCTP and RUDP have been designed as a tradeoff between TCP and UDP for streaming real-time videos over the IP networks. Flow control and error control schemes are crucial because of excessive delay and bursty packet loss nature of the wired and wireless packet networks. Rate control determines the sending rate of the video stream based on available network capacity and thus avoids congestion and loss of packets. Error control schemes can detect and correct data error and are therefore necessary for reliable video broadcasting.

These attributed are crucial for broadcasting video over a bandwidth constraint IP wired or wireless networks such as the cellular 3G/4G networks.

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