## WebTechnologies(La



## Lab1

#### **TheInternetProtocols**

Reema Mohammed Al-Khalifa 391202878

This lab session covers the usage of the Wireshark application to monitor and capture the outgoing and incoming packets from a network connection (WIFI, ethernet, etc.). Specifically, students should be able to analyze HTTP, HTTPS, TCP/IP, and UDP protocols using Wireshark, a network protocol analyzer, and draw conclusions.

## **Pre-lab Preparation:**

- 1. Review the basics and the structure of HTTP, TCP/IP, and UDP protocols,
- 2. Install Wireshark and ensure it is running on your computer,
- 3. Create an online, *publically accessible* Git repository to host and upload your work in the labs. We recommend you use GitHub or GitLab.

#### **Lab Activities:**

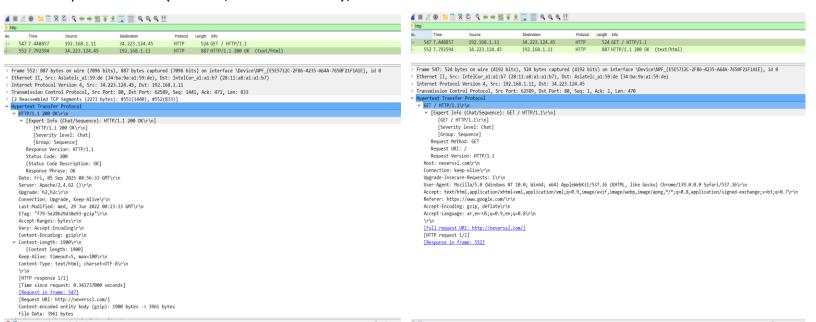
## **Part 1: Capturing HTTP Traffic.**

#### Task 1: Start Wireshark and capture packets.

- Step 1: Open Wireshark.
- Step 2: Select the network interface connected to the internet.
- Step 3: Click the "Start Capturing Packets" button (the shark fin icon).
- Step 4: Open your favorite web browser and navigate to (<a href="http://neverssl.com/">http://neverssl.com/</a>) website.
- Step 5: After the website has fully loaded, stop capturing packets by clicking the red stop button in Wireshark.

#### Task 2: Filter HTTP packets and analyze them.

- Step 1: In the filter bar, type http and press Enter. This filters out only the HTTP packets from the capture.
- Step 2: Select any HTTP packet to view its details.
- Step 3: Observe the HTTP request and response messages. Note the method (GET, POST), URL, response codes (200 OK, 404 Not Found), etc.



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- Part 2: Analyzing TCP/IP Traffic.
- **Task 1: Filter TCP packets**
- **Step 1:** Clear the previous filter and type TCP to focus on TCP packets.
- **Step 2:** Select a TCP packet related to your HTTP request/response.
- **Step 3:** Right-click on the packet and select "Follow" -> "TCP Stream".
- **Step 4:** This shows the entire conversation between the client and server.



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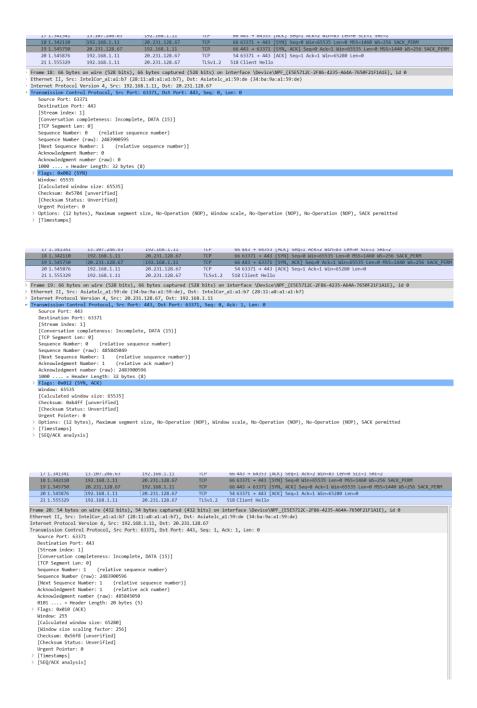
### Task 2: Analyze TCP handshake and investigate Data Transfer and Termination

**Step 1:** Find and select packets related to the TCP three-way handshake:

- o SYN: Initiates a connection.
- SYN-ACK: Acknowledges and responds to the SYN.
- ACK: Acknowledges the SYN-ACK and establishes the connection.

**Step 2:** Note the sequence and acknowledgment numbers. Screenshot and upload your image to your online git repository.

**Step 3:** Observe the data packets exchanged between the client and server. Take a screenshot and upload it to your online git repo.

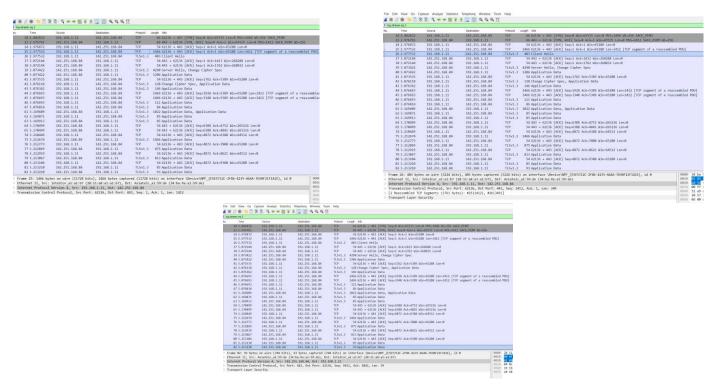


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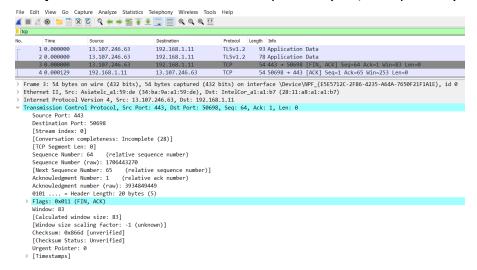


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### **Step 4:** Look at the TCP termination process (FIN, ACK packets).



#### Part 3: Capturing and Analyzing UDP Traffic

#### Task 1: Generate UDP traffic and capture packets

- **Step 1:** Open a network application that uses UDP (e.g., streaming video, VoIP software, or custom script).
- **Step 2:** Start the application to generate UDP traffic.
- **Step 3:** Start capturing packets in Wireshark while the UDP application is running.
- **Step 4:** After sufficient traffic is generated, stop capturing packets.

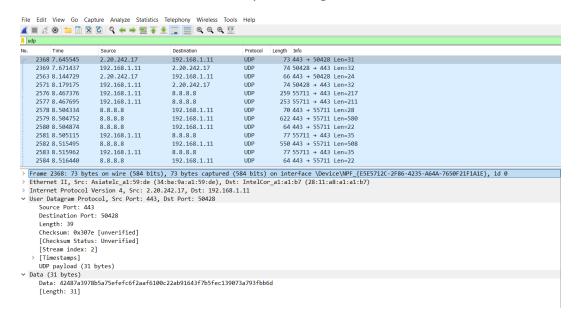
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- Task 2: Filter and analysis UDP Packets
- **Step 1:** In the filter bar, type UDP and press Enter.
- **Step 2:** This filters out only the UDP packets from the capture.
- **Step 3:** Select any UDP packet to view its details.
- **Step 4:** Observe the source and destination ports, length, and data.



**Step 5:** Compare the simplicity of UDP headers with TCP headers.

The table below summarizes the key differences between TCP and UDP headers based on the observed packet analysis.

Feature	TCP	UDP
Header Size	20 - 60 bytes (variable).	8 bytes (fixed)
Reliability	Reliable; ensures data delivery, ordering, and error checking.	Unreliable; no guarantee of delivery or order.
Connection	Connection-oriented; requires 3-way handshake (SYN, SYN-ACK, ACK).	Connectionless; no handshake required.
Sequence Number	Present; used to order data segments.	Not present.
Acknowledgment Number	Present; confirms received segments.	Not present.
Flags	Multiple flags (SYN, ACK, FIN, RST, PSH, URG) for controlling connection.	No flags
Simplicity	Complex due to reliability and connection management.	Simple and lightweight; fast transmission.

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# document), and upload it to your online git repo.

Task 1: Fill in the following table and provide reasons.

	TCP or UDP	Reasons
Reliability and Connection Establishment	ТСР	TCP is connection-oriented, establishing a reliable session via a three-way handshake (SYN, SYN-ACK, ACK) before data transfer. It ensures all packets are delivered by retransmitting lost or corrupted packets, making it highly reliable for applications like file transfers or web browsing. UDP is connectionless, sending packets without establishing a session, which can lead to packet loss and no retransmission.
Data Integrity and Ordering	ТСР	TCP ensures data integrity through checksums to detect errors and retransmits corrupted or lost packets based on acknowledgments (ACKs). It guarantees correct data ordering by assigning sequence numbers to packets and using ACKs to confirm their receipt in the proper order, allowing for reordering or retransmission of missing packets. UDP lacks these mechanisms, potentially leading to data loss or out-of-order delivery.

Task 2: Identify the use Cases and Performance of TCP and UDP.			
	ТСР	UDP	
Use cases	<ul> <li>File transfer (FTP, SFTP).</li> <li>Web browsing (HTTP/HTTPS).</li> <li>Email (SMTP, IMAP, POP3).</li> <li>Applications requiring high reliability and guaranteed data delivery.</li> </ul>	<ul> <li>Video and audio streaming.</li> <li>Online gaming.</li> <li>VoIP (voice over IP).</li> <li>Applications requiring speed and low latency rather than full reliability.</li> </ul>	
Performance	<ul> <li>Reliability: Ensures data delivery through acknowledgments, retransmissions, and error checking.</li> <li>Speed: Slower due to connection setup (three-way handshake), flow control, and congestion control mechanisms.</li> <li>Overhead: Higher due to headers (20 bytes minimum).</li> <li>Latency: Increased by retransmission delays and congestion avoidance, making it less suitable for real-time applications.</li> <li>Throughput: Limited by network congestion and round-trip time (RTT), but stable for large data transfers.</li> </ul>	<ul> <li>Reliability: No guarantees for delivery, order, or error correction; packets may be lost, duplicated, or arrive out of order.</li> <li>Speed: Faster due to no connection setup, acknowledgments, or retransmissions, making it ideal for low-latency needs.</li> <li>Overhead: Lower, with a minimal 8-byte header, reducing processing and bandwidth usage.</li> <li>Latency: Minimal, as it sends data without waiting for confirmation, suitable for real-time applications.</li> <li>Throughput: Potentially higher in uncongested networks, but performance degrades with packet loss in unreliable networks.</li> </ul>	