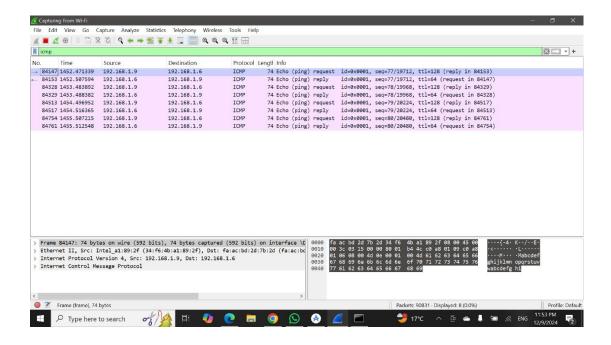
ICMP packets

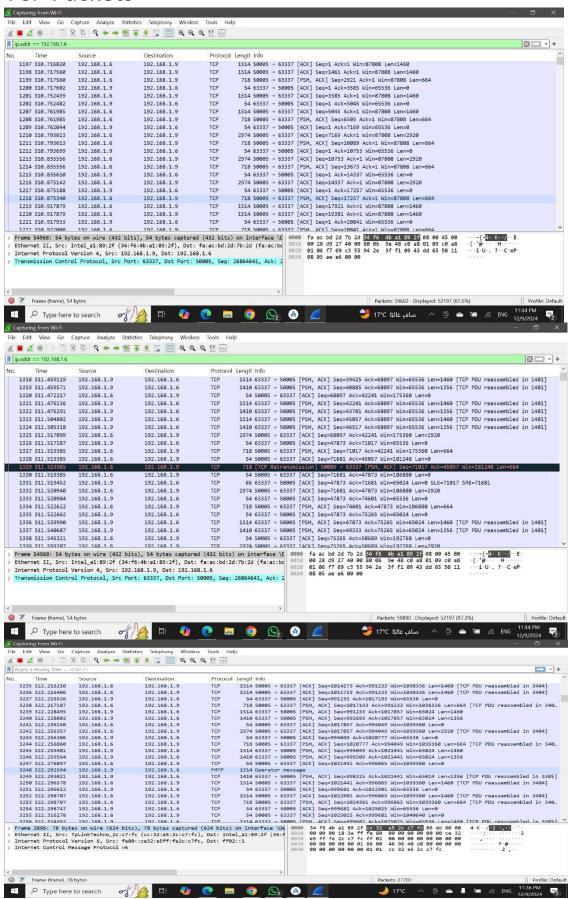


ICMP is used for pinging and make sure the connection is stable.

It is used for reporting errors and performing network diagnostics:

- A. Error Reporting:- Unreachable destination or time exceeded
- B. Diagnostic Tools:- Ping and trace route

TCP Packets



How the Code Works Server-Client Communication Setup:

Server: Creates a Server Socket on a specific port (50005) and waits for a client to connect. Once a client connects, it proceeds to handle communication.

Client: Connects to the server using its IP address and port (50005).

Audio Capture and Playback: Uses Audio Record to capture audio from the microphone. Uses Audio Track to play back received audio. Both are configured to use mono channels, a 44100 Hz sample rate, and PCM 16-bit encoding.

Audio Data Transmission:

Sending: Captured audio is read into a buffer and sent over the socket using the server-client connection.

Receiving: Audio data received from the socket is written to Audio Track for playback.

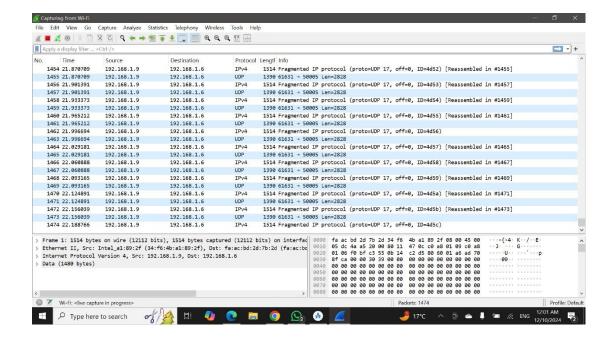
Threads for Parallel Processing: Separate threads handle sending and receiving audio data concurrently, ensuring continuous audio streaming.

Why It Doesn't Use SIP or RTP

No SIP Implementation:SIP is not used in this code because it doesn't perform any session initiation, modification, or termination signaling. The connection is directly established using raw sockets without a signaling protocol.

No RTP for Media Transport: RTP is a protocol designed for real-time audio/video streaming. However, the code uses raw TCP sockets to transfer audio data, which is simpler but less optimized for real-time communication compared to RTP.

Missing Protocol Handling: There is no SDP (Session Description Protocol) for negotiating media parameters, which is essential for RTP. RTP usually operates over UDP for lower latency, but this code uses TCP, which is less suited for real-time communication due to potential delays from re-transmissions.



Relationship Between RTP and UDP: RTP relies on UDP as its transport layer to provide real-time delivery of data. The additional RTP header (12 bytes minimum) gives context and management features for real-time media, which plain UDP lacks.

RTP over UDP in Practice: RTP packets are encapsulated inside UDP datagrams. UDP provides basic delivery, while RTP adds context for the media. Typically, RTP works alongside RTCP (RTP Control Protocol), which helps monitor and manage the RTP stream (e.g., reporting quality, handling synchronization).

Comparison between TCP and UDP:

1. Reliability

TCP (our Code): TCP is a connection-oriented protocol. It ensures reliable delivery of data through mechanisms like acknowledgments, re-transmissions, and sequencing. Data packets arrive in order, and missing or corrupted packets are resent. Suitable for applications where accuracy is critical, e.g., file transfers or web browsing.

UDP: UDP is a connection-less protocol. It does not guarantee the delivery, order, or integrity of data packets. Packets may be lost, arrive out of order, or be duplicated. Suitable for applications where speed is more critical than reliability, e.g., live audio/video streaming or gaming.

2. Latency

TCP: Higher latency because of connection establishment, acknowledgments, and re-transmission of lost packets. Suitable for scenarios where reliability outweighs speed, such as in the code example provided.

UDP: Lower latency since there is no connection setup or acknowledgment overhead. Ideal for real-time applications like voice or video calls where minor data loss is acceptable to maintain speed.

3. Data Transmission

TCP: Data is sent as a continuous stream, and the protocol ensures that the receiver gets the data in the correct order. Overhead from headers, handshakes, and re-transmissions can make TCP slower for real-time applications.

UDP: Data is sent in discrete packets (datagrams). No re-transmissions or guarantees, which makes it faster for time-sensitive tasks.