SIMATS ENGINEERING

ASSIGNMENT - 01

CSA0735 : COMPUTER NETWORKS

REGISTED NUMBER	NAME	QUESTIONS
192511172	SAMRAKSHINI. G	VOICE COMMUNICATION IN ENTERPRISES[VoIP]

SCENARIO : A FIRM USES VoIP FOR ALL INTERNAL COMMUNICATION.

QUESTIONS :

A) Compare traditional telephony vs VoIP on OSI layers.

OSI LAYER	TRADITIONAL TELEPHONY	VoIP VOICE OVER IP
7. APPLICATIO-N LAYER	It uses physical telephones, no digital application involved here	It uses softphones, IP phones or application like, skype, zoom, cisco
6. PRESENTA-TION LAYER	Not applicable [analog voice, no need for format conversion]	Converts audio into digital format [eg, codes like G.711, G.729 for compression]

OSI LAYER	TRADITIONAL TELEPHONY	VoIP VOICE OVER IP
5. SESSION LAYER	There is no session control protocol ; calls are circuit - switched and controlled by telephon -e switches.	It uses protocol like SIP [session initiation protocol] H.323 to manage sessions.
4. TRANSPORT LAYER	Here, CIRCUIT SWITCHING ensures continuous dedicated path, there is no concept of packets	It uses UDP [preferred due to low latency] or TCP for reliable control messages
3. NETWORK LAYER	Traditional telephony does not use packet - switch -ed network.	It uses IP for addressing and routing voice packets across networks.
2. DATA LINK LAYER	It uses dedicated lines [T1, E1 or analog], no frame based transmission	It uses Ethernet Wi-Fi or other Layer 2 technologi -es for framing.
1. PHYSICAL LAYER	Analog or data transmission over copper wires (twisted pair), fiber etc	Digital transmiss -ion over LAN cables, fiber optics or Wi-Fi

B) Recommend protocols for voice over IP.

1. **SIP** → SESSION INITIATION PROTOCOL

* Used for establishing, managing and terminating voice/video calls.
* Application layer protocol.

2. **RTP** → REAL-TIME TRANSPORT PROTOCOL

* Transmits real-time audio (and video) over IP network
* Ensures proper sequencing and timing of voice packets.

3. **RTCP** → RTP CONTROL PROTOCOL

* Works alongside RTP to monitor call quality.
* Provides feedback on delay, jitter and packet loss.

4. **UDP** → USER DATAGRAM PROTOCOL

* Used to transport protocol for RTP and RTCP

5. **H.323** - An alternative protocol suite for multimedia communication

6. **IP** → INTERNET PROTOCOL : Used for addressing and routing voice packets.

7. **DNS AND ENUM** → Used to map telephone number to IP addresses.

C) EXPLAIN JITTER BUFFER AND ITS NEED,

- **JITTER BUFFER** : It is a temporary storage at the receiving end of a VOIP system that collects incoming voice packets and arrange them in proper sequence before playback.

- **NEED FOR JITTER BUFFER** :
  - It ensures smooth and continuous audio by delaying playback slightly and buffering enough packets to compensate for delay variations.
  - It reduces choppy audio, echo and packet loss issues.

D) SUGGEST ERROR RECOVERY FOR VOICE PACKETS,

Voice packets are sensitive to delay and cannot be retransmitted. Error recovery is achieved using,

1. FEC → FORWARD ERROR CORRECTION : Sends extra data for reconstruction of lost packets.
2. PLC → PACKET LOSS CONCEALMENT : Fills in missing audio using previous data patterns.
3. QOS → QUALITY OF SERVICE : Prioritizes VoIP traffic to reduce loss and delay.
4. REDUNDANCY → Sends duplicate or partial voice packets to increase the chance of successful delivery.
5. ADAPTIVE CODECS → It adjust to changing network conditions to preserve audio quality.