VOIP Usage Analysis in a Remote Team Communication Scenario

Introduction

In the age of remote work and virtual collaboration, Voice over Internet Protocol (VOIP) has become a vital technology for business communication. This assignment analyzes the data Consumption, bandwidth needs, and Key protocols for a remote team of 20 users conducting daily 1-hour VoIP calls, using a 64 kbps code.

a) <u>Daily and Monthly Data Usage per User.</u>

Each user participates in a 1-hour VoIP call every day using a codec with a 64 Kbps (kilobits per second) bitrate.

<u>Daily Usage Calculation.</u>

* 64 Kbps = 64,000 bits per second
* 1 hour = 60 minutes = 3600 seconds

64,000 bps × 3600s = 230,400,000 bits

230,400,000 bits ÷ 8 = 28,800,000 bytes

<u>Monthly Usage</u>

* Assuming 30 working days.

28.8 MB/day × 30 = 864 MB/month

So, each user consumes approximarely 28.8 MB per day and 864 MB per month.

b) Total Bandwidth Required

For real-time communication, each user needs to send and receive data.

* Bandwidth per user = 64 kbps (send) + 64 Kbps (receive) = 128 Kbps

* 20 users simultaneously;

20×128 kbps = 2560 kbps = 2.56 Mbps

Therefore, the system requires a minimum of 2.56 Mbps symmetrical bandwidth to support all users concurrently with good call quality.

c) QoS (Quality of service) Policies for VoIP

VoIP is highly sensitive to delay, jitter, and packet loss. To ensure smooth and clear voice communication, the following QoS policies are recommended.

1. Traffic Prioritization:

* Assign high priority to VoIP pakets using DSCP (Differentiated services Code Point) marking.

2) Low latency Queuing (LLQ)

* Ensures that VoIP traffic is always served quickly, even during congestion.

3) Bandwidth Reservation:

* use tools like RSVP (Resource Reservation protocol) to allocate fixed bandwidth for voice.

4) Jitter Buffers:

* Temporarily store voice packets to handle delay variations.

5) Packet Shaping and policing:

* Manage non - VoIP traffic to prevent it.

a) Role of SIP in VoIP

SIP (Session initiation Protocol) is a key signalling protocol used in VoIP system. It is responsible for:

Function	Description
Session Setup	Initiates calls between endpoints (e.g., Phone to Phone)
User Location	Determines where the other party is located (IP address)
Session	Manages call Features like.

Management	hold, transfer, and re-invite.
Call Termination	Ends the call and releases network resources.

SIP works alongside RTP (Real-Time Transport Protocol) which carries the actual voice data during the call.

Conclusion

This assignment illustrates how a seemingly lightweight VoIP call can have significant bandwidth and quality requirements when scaled across a team. Proper calculation of data usage, accurate bandwidth Provisioning, and implementation of QoS techniques are crucial to ensuring reliable communication. The SIP protocol plays a vital in managing the entire life cycle of a VoIP session.