PERFORM BEYOND EXPECTATIONS

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TCP/IP & UDP Protocols FAQs

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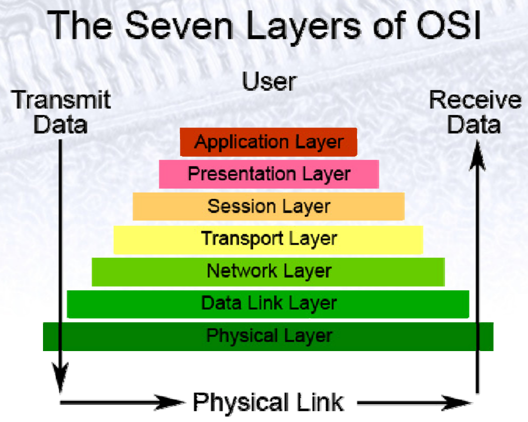

Converging IP applications primarily use either TCP/IP or UDP protocols to transmit voice, data and video simultaneously over IP networks. The use of both protocols simultaneously in a converging network increases the complexity of how network traffic is organized, transmitted and received.

Transmission Protocol Overview

- The Transport Layer of the Open Systems Interconnection (OSI) model communicates through host-to-host and via the LAN or remote networks
- The two main packet delivery protocols are TCP/IP (Transmission Control Protocol) and UDP (User Datagram Protocol)
- TCP/IP is a connection-oriented protocol in which delivery is guaranteed by receiver acknowledgement
- UDP, mainly used for time-sensitive applications uses a broadcast method in which delivery is not guaranteed.

What is TCP/IP?

TCP/IP (Transmission Control Protocol) provides end-to-end connectivity specifying how data should be formatted, addressed, transmitted, routed and received at the destination. TCP/IP is a connection-oriented protocol. This means that if all the components that make up a frame are not delivered together, the server will send a re-transmit request and wait until all the pieces are reconstructed to match what was originally transmitted. If the wait is too long, the connection will time-out.

What does TCP/IP guarantee in terms of quality of service?

TCP/IP provides both data integrity and delivery guarantee by re-transmitting until the receiver acknowledges the receipt of the frame (or packet).

What is UDP?

UDP (User Datagram Protocol) does not divide a message into packets and reassemble it at the other end. UDP is known as a “connectionless” protocol, and is simpler than TCP/IP. Multiple messages are sent or broadcasted as “chunks.” UDP is typically used for applications such as streaming media (audio, video, Voice over IP (VoIP), etc.), where on-time arrival is more important than reliability. Network traffic using UDP is given priority over TCP/IP, since on-time delivery is essential. However, without an error detecting code or automatic repeat request, UDP packets lost during transportation could result in dropped calls, choppy sound or excessive static for voice applications or corrupted frames for video applications.

What does UDP guarantee?

UDP provides some data integrity via a checksum but does not guarantee delivery. The checksum calculation is only to confirm the frame (or packet) received is intact. If not, there is no retransmit request and the frame (or packet) is discarded.

How are frame errors addressed by each protocol?

Both protocols will handle frame errors differently. A frame error with UDP, will mean the frame is dropped forever and whatever was inside the frame will not be received. In TCP/IP, re-transmit requests will be sent in hopes of receiving an error-free transmission – but this process can result in sluggish network performance.

How can I make sure that my network is able to manage the variety of protocols in use today and into the future?

Berk-Tek's products are subjected to real world environmental conditions and demonstrate reliable and robust performance. We use sophisticated analytical tools to measure the performance of networks using our industry leading LANmark™-1000 and LANmark™-2000 products. Specifically, LANmark™-1000 and LANmark™-2000 are characterized using Mean Opinion Score (MOS), Frame Error Rate (FER), and Media Loss Rate (MLR). MOS tests for VoIP quality, which uses UDP, FER tests for data integrity which employs TCP/IP, and MLR tests for IP video quality and can use either TCP/IP or UDP depending on the type of video. This robust and varied testing provides IT managers with the confidence to leverage IP convergence without worrying about degradation to their networks.

The below table is a quick reference guide to compare and contrast the two different protocols.

ITEM	TCP/IP	UDP
Connection	Connection-oriented protocol - Established through a "handshake" before data can be sent	Connectionless protocol - Packets are sent individually and transported on top of IP.
Packet Entity	Segments	Datagram
Reliability	Reliable byte stream - messaging is managed and acknowledged. TCP does error checking and has recovery methods as well as traffic congestion control	Error checking through "checksum" but no recovery options.
Ordering	Transmissions are sent in sequence.	Time sensitive and time preferential. No sequencing or ordering of messages.
Recovery Methods	Lost or discarded packets are resent	No recovery
Applications	High-reliability and less critical transmission time - World Wide Web, file transfer, e-mail	Fast timing - Real time streaming protocol – Multicasting - Voice over IP, online gaming, video streaming, IPTV
Priority	Low priority since timing is not as important and delivery is confirmed	First priority due to application's time sensitivity