Video Lecture 3: Voice over IP Service

7COM1030 – Multicast and Multimedia Networking

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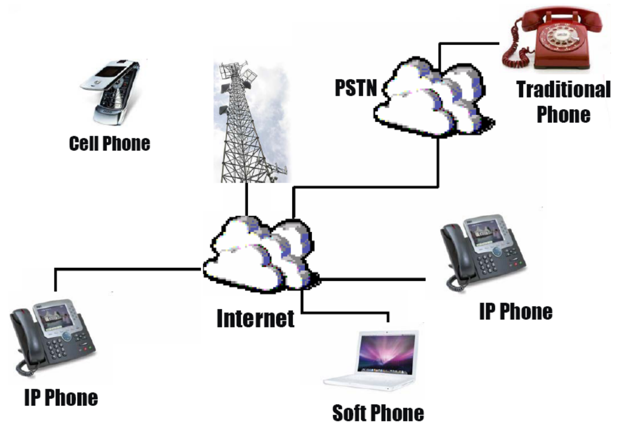
What is Voice over IP?



PSTN – Public Switched Telephone Network

Why use VoIP?

- Cost savings
- Integrated data and voice networks
- Device interoperability using standards-based protocols
- Flexibility in driving new services

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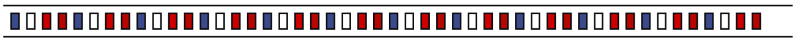
Traditional Voice vs. VoIP

- A traditional T1 can carry **24** telephone calls simultaneously

T1 = 1544 kbps, DS0 = 64 kbps,

1544 / 64 = 24 DS0 per T1

- With VoIP, a T1 could carry **64** calls simultaneously



G.729 8kbps compression, 20 msec frame size = 24 kbps

1544 / 24 = 64 calls per T1

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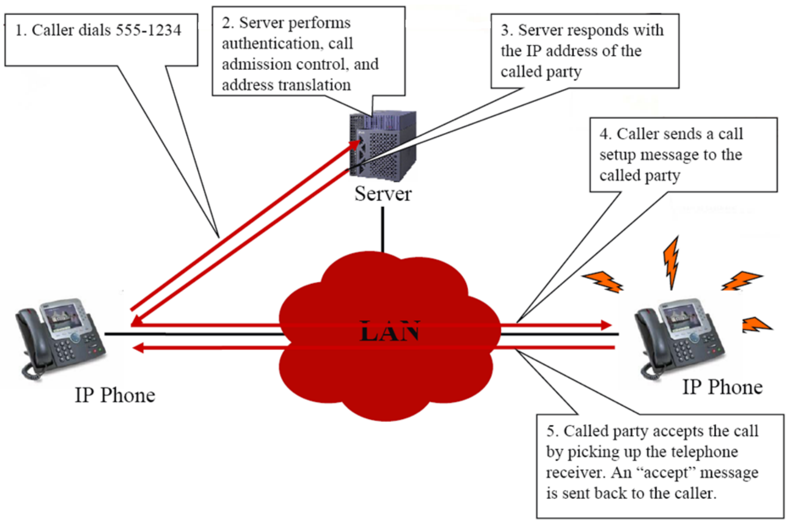
Call Setup (IP-to-IP)



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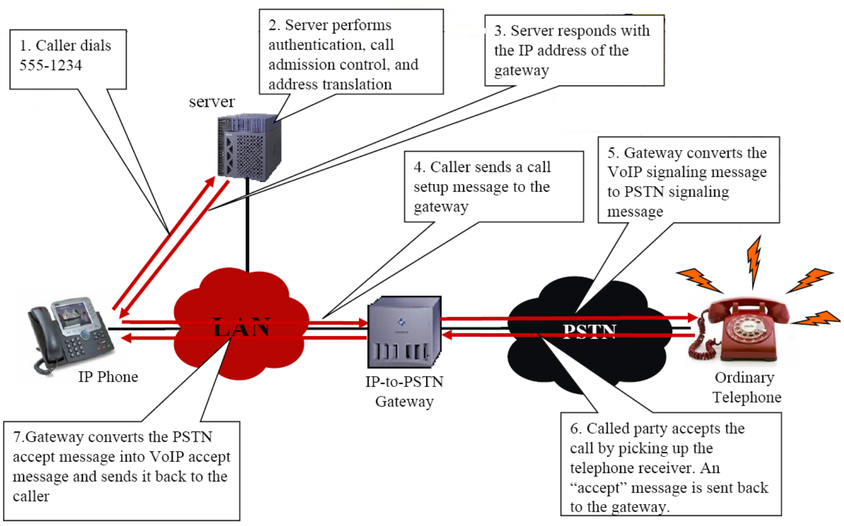
Call Setup (IP-to-PSTN)



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Data Plane Protocols

- RTP – Real-time Transport Protocol
  - Maintains sequence of packets
  - Time stamping
  - Delivery monitoring
  - Minimal session control
- Normally RTP is built over UDP instead of TCP, as the application requires speedy delivery of information and can tolerate some packet loss.

Think why TCP is not suitable for VoIP?

Measurement Criteria of VoIP Performance

- **Jittering** is the variability in the delay which is computed as expected arrival time minus actual arrival time. It must be less than 30ms. De-jitter buffer helps fix the problem, but adds to the overall delay.
- **Packet Loss** measures the percentage of the dropped packets which should be less than 1%.
- **MOS** (Mean Opinion Score), ITUT-P.800 standard defines MOS as a subjective metric which estimates the user satisfaction by means of a score which varies from 1.0 (poor) to 5.0 (best). The minimum MOS should be maintained at level 3 to achieve an acceptable performance.
- **ETE Delay** (end-to-end) is the time that elapses between when an utterance is spoken and when it is played back at the receiver, which must be less than 150 ms for real-time conversations.

Parameters	Acceptable Level
Delay	$\leq$ 150ms
Jitter	$\leq$ 30ms
Packet Loss	$\leq$ 1%
MOS	$\geq$ 3

Nature of Different Routing Protocols

- Routing is an essential data networking function that provides an efficient real-time data delivery required by VoIP.
- Best-effort networks leverage IGP technologies to determine paths for routing packets between hosts.

	RIP	OSPF	EIGRP
NATURE	DV	LS	Hybrid
SCALE	Small networks	Enterprise networks	Medium
ROUTING	Classful Routing loop counter mechanism	Classless	Classless 100% loop free
METRICS	Number of hops	The inverse of the bandwidth of links	Available bandwidth, delay, load, MTU and the link reliability
DISCOVERY AND UPDATES	Periodical updates (broadcast)	DR multicasts whenever changes are made	DUAL Multicast Incremental update
FAILURE RECOVERY	Slow convergence	Generally faster than RIP [16]	DUAL algorithm
LOAD BALANCING	Only supported on equal-cost paths	Supports 6 equal-cost paths, but difficult to implement	Supports 6 unequal paths, but commonly ignored due to its complexity and instability.

Performances of Different Routing Protocols

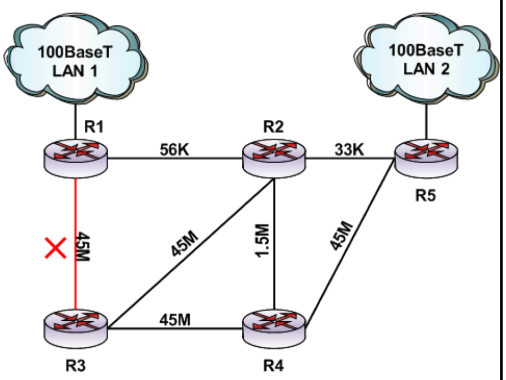
- RIP carries out low-efficient routing in the network with a bottleneck transmission link as it does not take bandwidth into consideration.
- In contrast, RIP and EIGRP perform with excellence as they are devoted to computing the fastest possible route. With the same network specifications it is likely that EIGRP and OSPF have chosen the same route for the VoIP application.
- Network failure can affect the performance of both EIGRP and OSPF. OSPF can act consistently throughout the failure recovery procedure, while EIGRP may need special consideration for service to continue.

Failure Recovery: OSPF vs. EIGRP

- OSPF updates the routing table upon network failure to re-calculate a new route, and does not alter the route for any existing traffic stream as long as there are no congestions or other new problems in its chosen route.
- EIGRP uses the Diffusing Update Algorithm (DUAL) to determine the routes. DUAL enables EIGRP routers to determine whether a path advertised by a neighbour is looped or loop-free. The DUAL algorithm is not as efficient as OSPF when no feasible successor is found.

Case Study

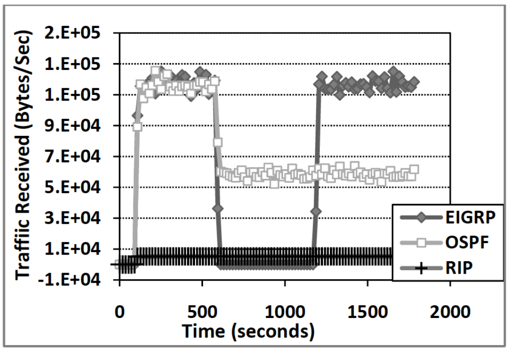
- A few routers are connecting two office branch networks that have 100BaseT specifications.
- Two bottleneck channels: 56K date rate between R1 and R2, and 33k data rate between R2 and R5.



The link between R1 and R3 failed after 10 minutes since the system started up, and recovered after another 10 minutes.

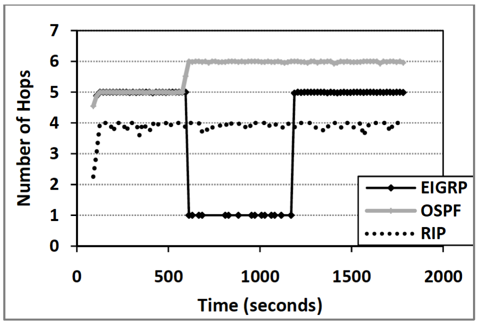
Throughput

- RIP chooses the route with bottleneck links, leading to insufficient data delivery.
- EIGRP suffers from the network failure.



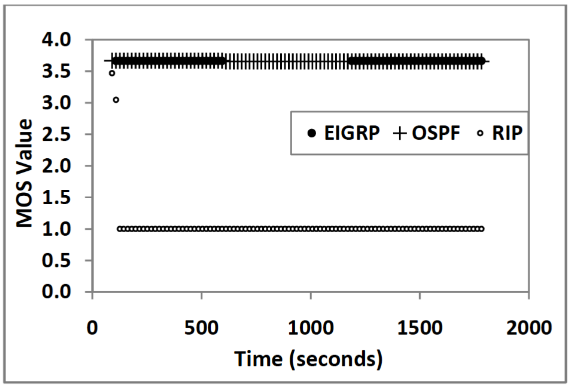
Number of Hops

- Upon the point failure, OSPF has chosen an alternative route to continue the date delivery process, and sticks to that route even after the failure is recovered as long as there are no congestions or other problems in its chosen route.



MOS Value

- OSPF yields consistent and acceptable performance, even with a failure recovery process involved.



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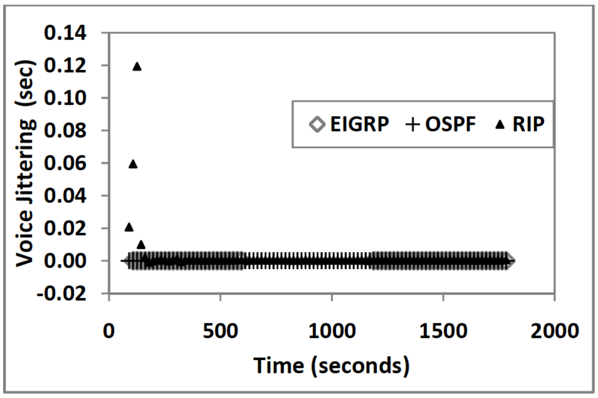
Voice Jittering

- RIP encounters serious jittering in the beginning of the VoIP conversation.



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Home Reading

- Studynet\Teaching Resources\Application Part
  - **VoIP Performance over Different Interior Gateway Protocols**

Questions?

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