Voice Over IP

An Overview

Reading

- There are literally thousands of references on VoIP due to the large market and many available products.
- Here is a short tutorial:

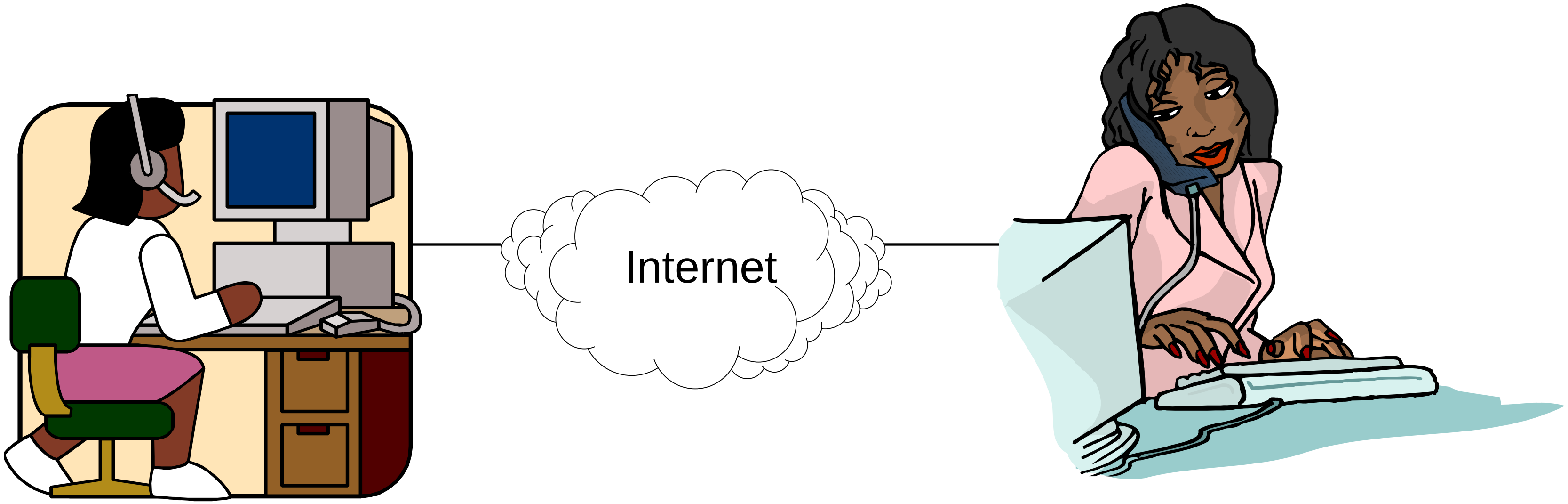
  `http://www.iec.org/tutorials/int_tele/`
- Here is a good collection of links:

  `http://www.iptelephony.org/frame/technology.html`
- M. Hamdi et al, "Voice Service Interworking for PSTN and IP Networks," *IEEE Communications Magazine*, Vol.37, No. 5, May 1999.
- Virtually all VoIP implementations use H.323
  - You're all experts on it by now

What is it, where did it start?

- Voice Over IP refers to using the IP network infrastructure for voice communications (and, more recently, for fax as well).
- First product appeared in February of 1995:
  - Internet Phone Software by Vocaltec, Inc.
  - Required a 486/33 MHz PC with sound card, microphone, speakers and modem.
  - Software compressed the voice and sent it as IP packets.
  - Used to make "free" long-distance calls
- Other products and free software soon followed.

Scenario 1: PC to PC



- Issues:
  - Both PCs need to have the same software
  - Protocol compatibility
  - Network issues

Technical Issues (Scenario 1)

- **End-to-End Delay:** causes the following two problems:
  - Echo: becomes an issue if the round-trip delay is over 50 milliseconds
  - Talker overlap: becomes an issue if the one-way delay is greater than 250 milliseconds.
- **Jitter:** since packets need to be played in a steady rate, the traffic will have to be buffered to compensate for the worst-case jitter.
  - Jitter adds to end-to-end delay.

Delay Budget Example

- Normal Phone: 10 ms/1000 miles $\Rightarrow$ 30 ms coast-to-coast
- G.729: 10 ms to serialize the frame + 5 ms look ahead + 10 ms computation = 25 ms one way algorithmic delay
- G.723.1 = 100 ms one-way algorithmic delay
- Jitter buffer = 40-60 ms
- Data processing and sub-optimal implementations in the PC $\Rightarrow$ 400 ms
- Conclusion: unless the system is properly engineered, it is very easy to exceed the maximum delay limits.

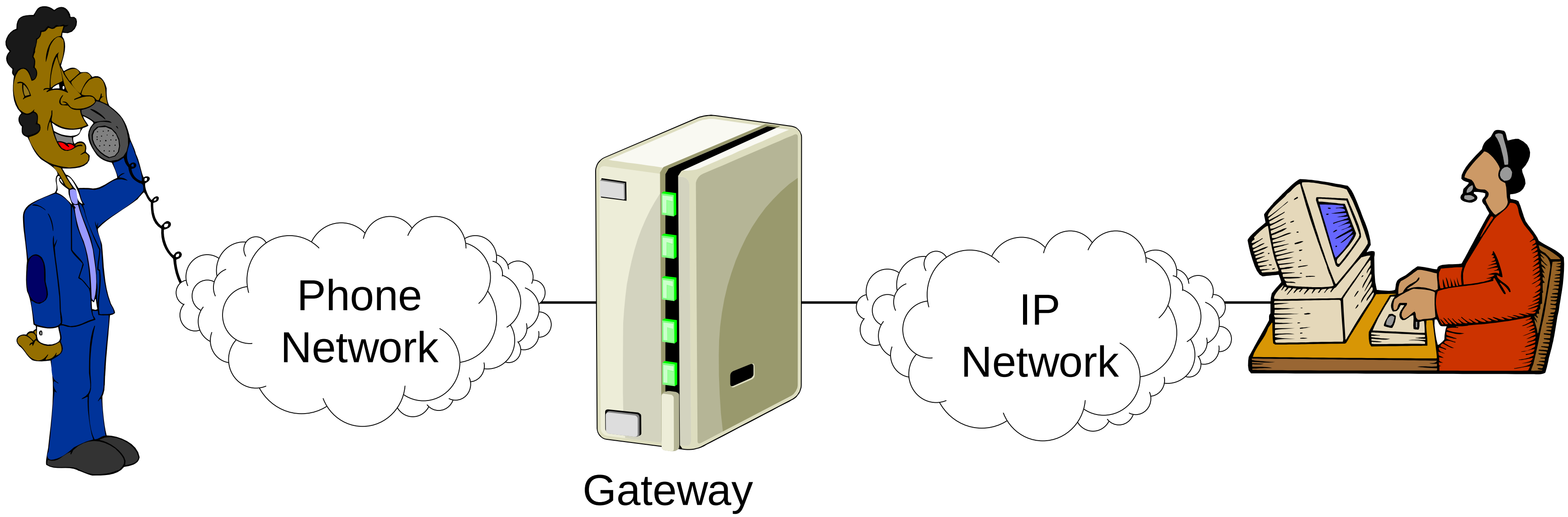
Scenario 1: Packet Loss

- IP networks are best-effort.
- Packets can be dropped due to congestion, or can arrive out-of-order due to routing over multiple paths.
- Some error concealment is possible:
  - Waveform interpolation for lost packets
  - Replace by silence
- Packet loss over 10% is unacceptable.
- Quality of Service, bandwidth allocation is a requirement to achieve voice toll quality.

Scenario 1: Overhead

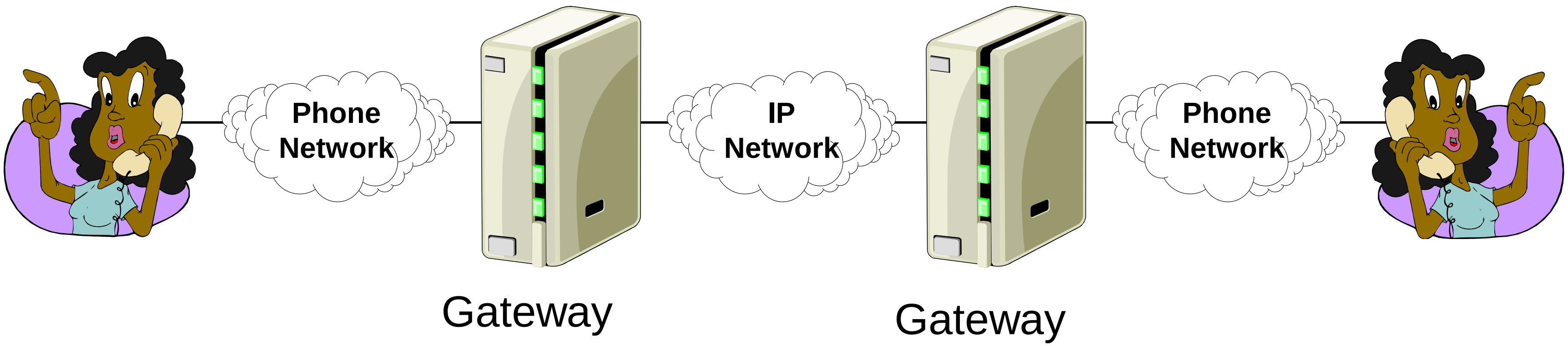
- Overhead can be an important factor in packet voice communications:
  - Increasing the packet size decreases the overhead
  - Decreasing the packet size decreases the latency and the packet loss susceptibility
- Overhead analysis:
  - 8 kb/s voice compression ⇒ 1 byte/millisecond
  - Target 20 ms latency ⇒ 20-byte packets
  - Overhead: IP (20), UDP (8), RTP (12) ⇒ 40 bytes
  - Total Overhead: 40/(20+40) ⇒ **67%**
  - However, packet networks allow statistical multiplexing!

Scenario 2: PC to Phone



- A Gateway is needed to connect the PSTN to the IP network:
  - Signaling conversion
  - Format conversion

Scenario 3: Phone to Phone



- Gateways will connect the phone network to the IP network.
- The IP Network can be a dedicated backbone or intranet (to provide guaranteed QoS) or can be the Internet (no guarantees …)
- The phone network can be a company PBX or carrier switches

Standards

- Virtually all VoIP implementations are based on the H.323 series of standards.
- The complexity of the standards leads to poor interoperability between vendors.
- The IMTC (International Multimedia Telecommunications Consortium), an industry group, published in 1998 the VoIP Interoperability Implementation Agreement (IA 1.0), which defined a baseline for implementations to interoperate.
- There are some alternate IETF standards also being considered/used in some areas.

The IETF Side

- A number of IETF standards have been adopted by the ITU-T as part of the H.323 series:
  - RTP; RTCP; support for RSVP; DTMF RTP Profile
  - H.235 suggests the use of IPSec to secure the H.225 negotiation.
  - The H.248 protocol used in the gateway decomposition for the communication between the MGC and the MG is a joint development with the IETF, and has been published as RFC 3015, "Megaco Protocol 1.0", November 2000.
- Directory services are still an open area
  - LDAP (RFC 2251) is a possibility.

Alternative: SIP/SDP

- The Session Initiation Protocol (SIP, RFC 2543) has been proposed as an alternative to H.323.
- SIP is capable of negotiating a call.
- SDP is used to describe capabilities.
- Because of their textual nature, SIP/SDP are easier to implement than the H.323 binary protocols.

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- Media still runs over RTP.

Comparison SIP - H.323 Functionality

	H.323 v1	H.323 v2	H.323 v3	SIP
Call Control Services:				
Call Hold	No	Yes	Yes	Yes
Call Transfer	No	Yes	Yes	Yes
Call Forwarding	No	Yes	Yes	Yes
Call Waiting	No	Yes	Yes	Yes
Call Park and Pickup	No	No	Yes	Yes
Message Waiting Indication	No	No	Yes	No
Call Completion on Busy Subscriber	No	No	Yes	Yes
Advanced Features:				
Third Party Control	No	No	No	Yes
Conference	Yes	Yes	Yes	Yes
Conference Control	Yes	Yes	Yes	No
Conference out of Consultation	No	Yes	Yes	Yes
Capability Exchange	Better	Better	Better	Yes
Web-Integration	Yes	Yes	Yes	Better

Source: EE-384B invited lecture by I. Dalgic, 1999

Comparison SIP - H.323 QoS and Management

	H.323v1	H.323v2	H.323 v3	SIP
Call Setup Delay	6~7 RT	3~4 RT	2~3 RT	2~3 RT
Call Control Reliability:				
Fault Detection	Yes	Yes	Yes	Yes
Fault Tolerance	No	No	Yes	Some
QoS Management:				
Admission Control	Yes	Yes	Yes	Limited
Policy Control	Yes	Yes	Yes	No
MIB	No	Yes	Yes	No
QoS Signaling	No	No	Yes?	No