Re: ip phone design considerations

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In article <3360d677.0406270110.42d3df59@posting.google.com>,
Apparatus <apparatus.home@lycos.com> wrote:
>I am planning on designing an IP phone for a student project. I would
>like some advice on details that I need to take into consideration.
[...]
>4) Is UDP a reliable enough transmit method? Should I add a 100ms
>buffer for frame delays? Should I repeat the last frame if a frame is
>omitted?

"Reliable enough" depends on the specific network that's carrying the IP traffic. As others have pointed out, if you're just going over one ethernet segment, it's pretty reliable; if you're transmitting to another continent, expect to have packets dropped regularly.

Rather than repeating the last piece of sound if you have a gap in the data, it's probably better to fill it with silence. This is less distracting to a human listener (expecially if you have a whole bunch of gaps in a row, i.e., the transmitter has stopped).

In addition to lost packets, you might also have to deal with:
- packets arriving out of order
- packets getting duplicated and arriving twice (this is rare)
- the transmitter and receiver's sample clocks being of slightly different frequencies
- the network delay between the transmitter and the receiver can change over time

The solutions to these are left as an exercise to the student :−) but you'll find lots of discussion about it on the net. In general, you end up having a tradeoff between solving these problems and introducing more end−to−end delay. The longer the delay is, the harder it is to hold a conversation over the link. Even fairly short delays can have a subliminal effect on how the person at one end perceives the other person's emotions.

(Re TCP −−− the thing to realize about TCP is that a TCP packet (that is, an IP packet carrying a piece of a TCP session) is no more reliable than a UDP packet. TCP simply notices when a packet

has been dropped and [oversimplification] requests that it be
retransmitted. In the meantime, the receiver doesn't see any data
−−− the connection pauses for a bit. This is the right behavior
for, e.g., a file transfer, but it's not what you want for a real−time
application.)

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