ip phone design considerations

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Hello,

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.

My basic thought is this: Use an ADC to take 12−bit samples of the microphone at 8kHz. Next encode the samples into u−law g.711 PCM. Next transmit over UDP packet. Next decode from u−law. Next send to 12−bit DAC connected to PGA or op amp, then to speaker. Note that I am planning on using a DSP on both ends, probably a Microchip dsPIC due to the affordability of the development tools.

The concerns I have are mainly on the analog ends.

1) How much of a problem will noise be with this scheme? Should I add a BPF before encoding the remove frequencies outside 400Hz to 4kHz? How else should I remove noise?

2) Is interpolation on the receiving end needed to achieve a toll quality signal? Can this interpolation be a simple capacitor or coil? Will the speaker be a good enough interpolator?

3) In general, how is a microphone connected to a ADC? A DAC to a speaker?

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?

5) What else should I consider? Any other suggestions?

Cheers.