Test Plan

Requirements	How to set up the test	What to test	Measures	Results (yes/no)
Audio should be digitized at a sample rate of 48 ksps	Look at sample rate in code	Is the audio digitized at rate of 48 kbps?	Rate = 48000	Yes
Each sample should be 16 bit value	Look at bit value in code	Is each sample a 16 bit value?	Format = pyaudio.paInt16	Yes
The total data transfer rate should not exceed a maximum of 10 kbps (kilo bits per second) between peers.	Look at Transfer Rate	Does the transfer rate exceed 10 kbps	8000-15000 avg	Not conistantly
The two devices are identical.	Open software on two computers on the same network	Does the software work on both devices?		Yes
Develop a device that meets the latency performance specifica-tions using two nodes on the CoE network (not localhost).	Setup up the connection, Look at the how the devices are connected	Is it over the network? (not on localhost)	Over university ethernet	Yes
The data transmission should occur through a tcp or udpsocket over the network.	Look at how data is transfered	Is the transfer method udp/tcp via sockets?	Socket programming(TCP)	Yes
Only a single channel is required	Look at the channel/signal	Is there only one channel/signal?	channels = 1	Yes
Develop an application that simlutaneously transmits and receives audio from one client to another with no latency of less than 300 ms	Have a conversation	Is audio transmitted ?		Yes
	Look at what is sent/received	Is audio received?		Yes
	Look when both processes are happening	Does this happen simultaneously?		Yes
	Look at latency/measure the latency	Is the delay less than 300 ms?	Best is .1 seconds	Yes
Design the necessary software to compress and recover the audio signal	Look at what is sent when talking	Is the audio compressed?	packets are smaller	Yes
	Look at what is received / audio played.	Is the audio recovered?		Yes
Design the hardware and software necessary to play audio received via a network connection over a speaker or set of headphones.	Plug in headphones/setup speakers. Continue conversation	Is audio heard via headphone/speakers?		Yes
Jitter in the arrival time of audio data packets should be handled through the use of a simple (fixed) jitter buffer	Look at the receiving code	Is there a simple fixed jitter buffer?		Yes
	While continuing the conversation	Is the conversation relatively normal? (no out of order or gaps in audio)	Echo, but no gaps	Yes
The conversation should last for at least a minute	Continue having the conversation	Can the conversation last for at least a minute?		Yes