After the simulation of 2-element array, next I am going to do is to figure out an algorithm for 64-elment array to achieve the beam forming.

The algorithm I introduce is the Delay-sum beam forming.

Delay-sum beam forming is a signal processing technique in which the outputs from an array of microphones are time delayed so that when they are summed together, a particular portion of the sound field is amplified over other unwanted or interfering sources. Several microphone array configurations exist, but a linear array has been chosen for less processing complexities and its effectiveness over a 180° field. Figure 1illustrates this setup with a sound source S placed in the field at an angle θ.

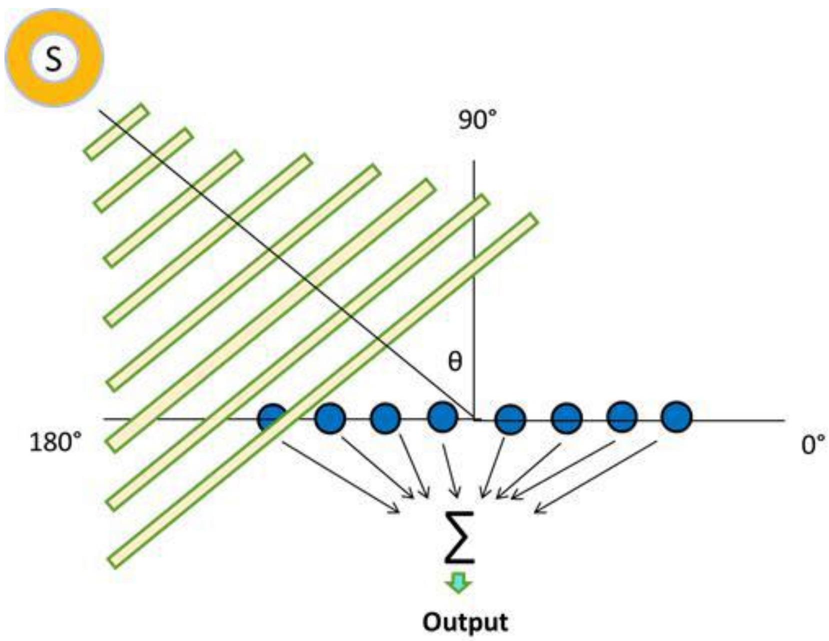


Fig. 2: Linear Beam forming array illustrating sound source in field

If we observe the sound waves emitted from source S, it can be seen the microphone furthest to the left captures the sound waves first. The adjacent microphones will receive the same signal, but with a slight time delay due to the additional distance sound waves must travel to get to the next microphone. When we summate the outputs of the individual microphones, we get (1)

$$\sum Output = S(t) + S(t - \Delta d_1) + S(t - \Delta d_2) + \cdots + S(t - \Delta d_n) \qquad (1)$$

S(t) is the wave equation representing the signal emitted from sound source S. The first microphone has the output S(t), and each subsequent microphone has a time delay Δdn, where n represents the microphone index (leftmost microphone has n=0, and the rightmost microphone has n=(number of microphones – 1)). Taking the Fourier transform of each term, this output can be represented in the frequency domain as a series of complex valued functions as shown (2), in which f is frequency.

$$\sum Output = S(f) + S(f)e^{-j2\pi(\Delta d_1)} + S(f)e^{-j2\pi(\Delta d_2)} + \cdots + S(f)e^{-j2\pi(d_n)} \tag{2}$$

In (2), it can be seen that there is a maximum magnitude if the time delays (Δd1 to Δdn) are equal to 0. This is the principle of delay-sum beam forming. Let's say we want to amplify the received signal from source S at angle θ in Figure 2. Knowing the parameters of our array, we can calculate the time delays that sound waves emitted from a source at angle θ will cause (calculation methods detailed in Section a). Let's call this calculated delay ΔdTAn, in which the term TA denotes these are calculated at our "target angle" θ, and n, or the microphone index, refers to the microphone to have this time delay. Applying these delays to (2), we get (3) below.

$$\sum Output = S(f) + S(f)e^{-j2\pi(\Delta d_1 - \Delta d_{TA1})} + \cdots + S(f)e^{-j2\pi(\Delta d_n - \Delta d_{TAn})} \tag{3}$$

If our calculations of ΔdTAn are accurate, they should be equal to the Δdn seen in each term. This simplifies our expression to (4) since (Δdn - ΔdTAn) will equal 0. The result is the sound waves heard from source S, now expressed in the time domain as the wave equation S(t), will have its magnitude multiplied by a factor of (n+1), the number of microphones in our array. This will amplify the signal S(t) in our summated output, but in order to localize this source, it must be amplified considerably more than other sound sources that appear in our field. This quality is known as the spatial resolution of our design, which can be more simply characterized as our "beamwidth."