Parte 2

Generar el archivo codificado con PCM 64 con ley A

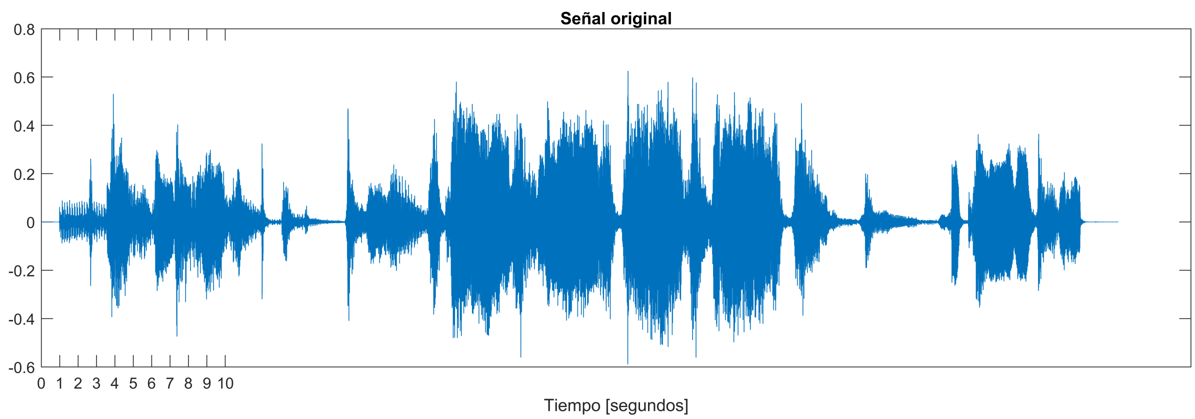
Leo el audio

https://www.mathworks.com/help/matlab/ref/audioread.html

```
[y_, Fs_] = audioread('recording.mp3');
Fs_
```

```
Fs_ = 44100
```

```
h = figure();
plot(y_)
xlabel("Tiempo [segundos]")
xticks([0:8000:8000*10])
xticklabels(["0","1","2","3","4","5","6","7","8","9","10"])
title("Señal original")
set(h,'Units','normalized','Position',[0 0 1 .5]);
```



Downsapling a 8kHz

https://www.mathworks.com/help/signal/ug/changing-signal-sample-rate.html

```
Fs = 8000
```

```
Fs = 8000
```

```
[P, Q] = rat(Fs/Fs_);
y = resample(y_, P, Q);
```

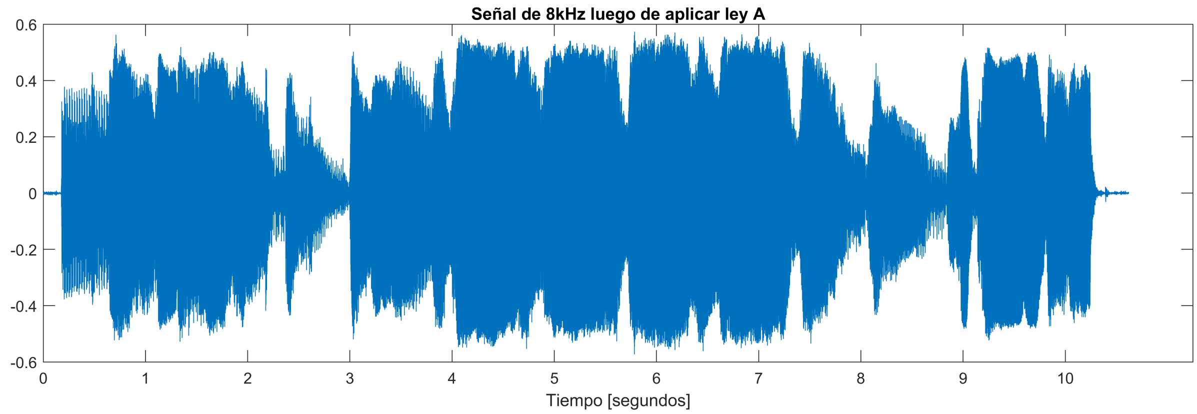
Aplicamos Ley A:

https://es.wikipedia.org/wiki/Ley_A

https://www.mathworks.com/help/comm/ref/compand.html

```
compressed = compand(y,87.6,max(y),'A/compressor');
plot(compressed)
xlabel("Tiempo [segundos]")
```

```
xticks([0:8000:8000*10])
xticklabels(["0","1","2","3","4","5","6","7","8","9","10"])
title("Señal de 8kHz luego de aplicar ley A")
```



Cuantizamos la señal

https://www.mathworks.com/help/comm/ref/quantiz.html

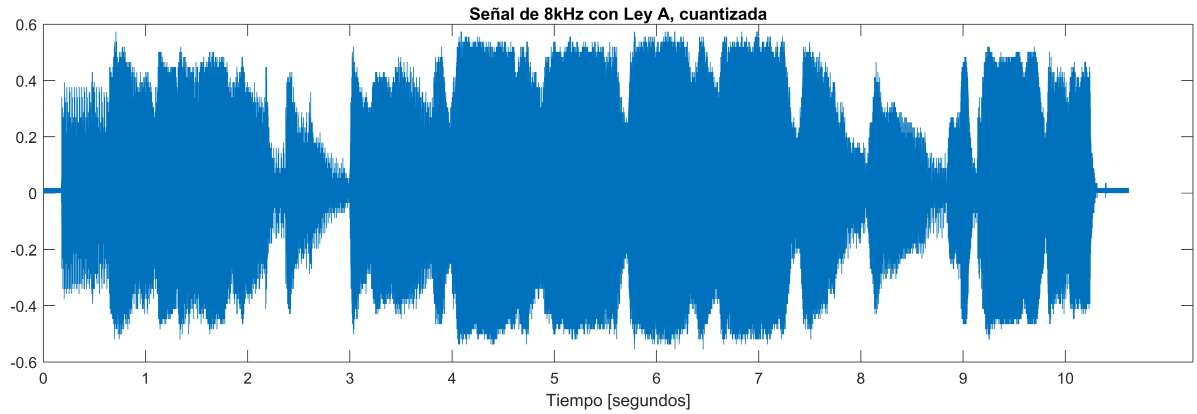
```
levels = 64
```

```
levels = 64
```

```
q = max(y)/(levels/2)
```

```
q = 0.0179
```

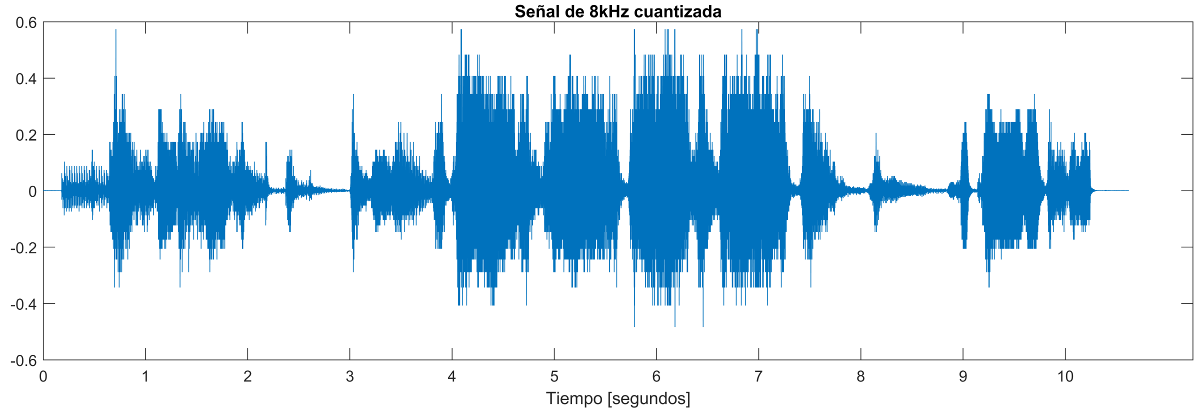
```
partition = -max(y):q:max(y)-q;
codebook = -max(y):q:max(y);
[index, quants, distor] = quantiz(compressed, partition, codebook);

plot(quants)
xlabel("Tiempo [segundos]")
xticks([0:8000:8000*10])
xticklabels(["0","1","2","3","4","5","6","7","8","9","10"])
title("Señal de 8kHz con Ley A, cuantizada")
```



Recupero la señal

```
expanded = compand(quants, 87.6, max(y), 'A/expander');
```

```
plot(expanded)
xlabel("Tiempo [segundos]")
xticks([0:8000:8000*10])
xticklabels(["0","1","2","3","4","5","6","7","8","9","10"])
title("Señal de 8kHz cuantizada")
```



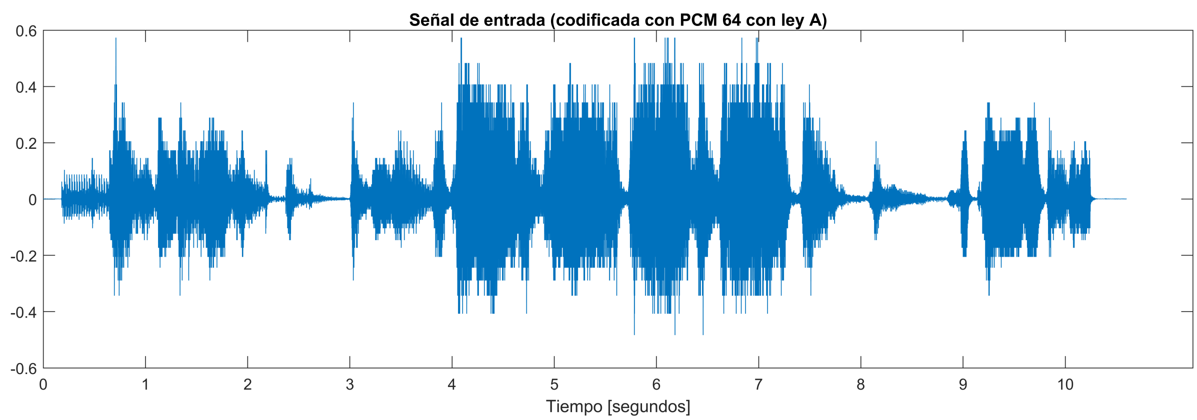
Guardo el audio

```
samples = Fs*20e-3;
% guardo un múltiplo de samples
len = floor(length(expanded)/samples) * samples;
audiowrite('audioPCM64.wav', expanded(1:len), Fs)
```

Agregar ruido

Se lee la señal previamente generada

```
[signal, Fs] = audioread('audioPCM64.wav');
plot(signal)
xlabel("Tiempo [segundos]")
xticks([0:8000:8000*10])
xticklabels(["0","1","2","3","4","5","6","7","8","9","10"])
title("Señal de entrada (codificada con PCM 64 con ley A)")
```



Generamos ruido blanco con varianza σ^2

https://www.mathworks.com/help/comm/ref/wgn.html

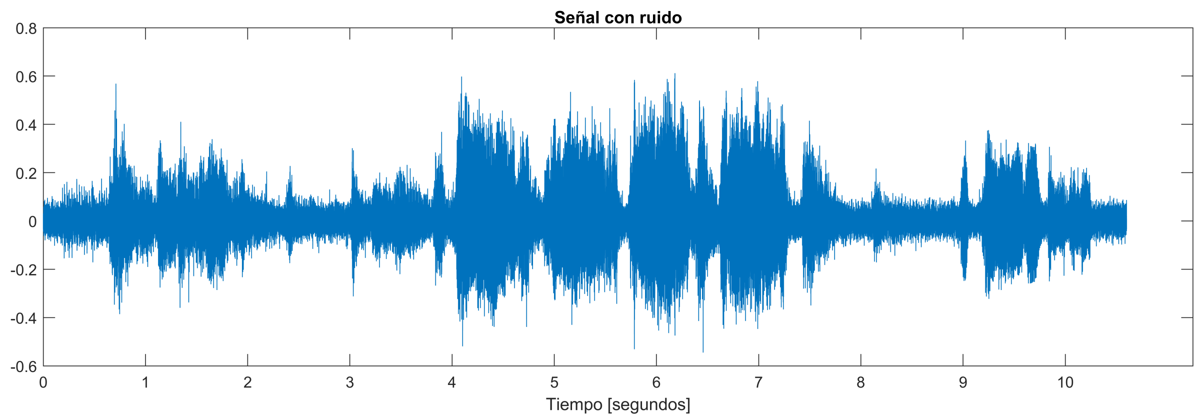
```
varianza = 1e-3;
power_dBW = 10*log10(varianza)
```

```
power_dBW = -30
```

```
nsamples = length(signal);
wgn_samples = wgn(nsamples, 1, power_dBW);
```

Lo agregamos a la señal de audio

```
signal_with_wgn = signal + wgn_samples;
plot(signal_with_wgn)
xlabel("Tiempo [segundos]")
xticks([0:8000:8000*10])
xticklabels(["0","1","2","3","4","5","6","7","8","9","10"])
title("Señal con ruido")
```




```
audiowrite('audioPCM64_con_ruido.wav', signal_with_wgn, Fs)
```

Predictor lineal óptimo

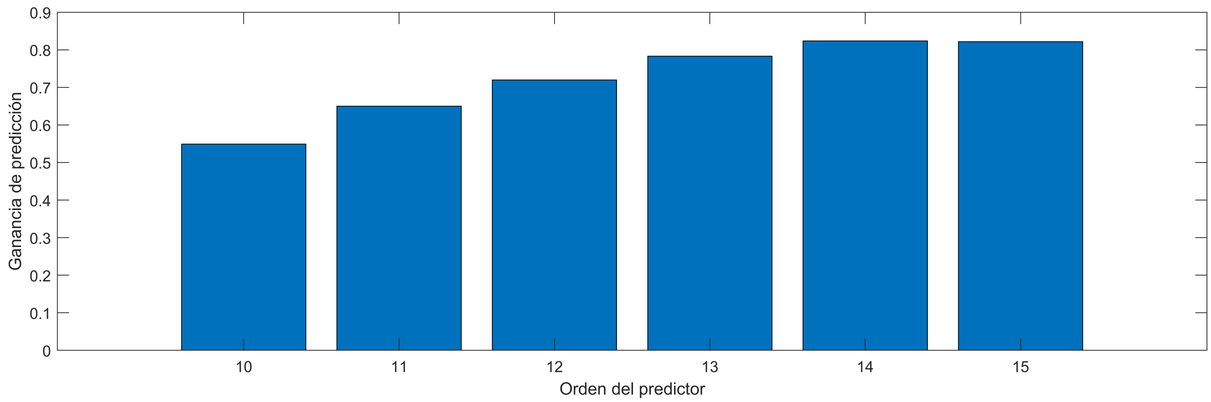
https://www.mathworks.com/help/econ/autocorr.html

```
orders = 10:15;
gps = [];
signal_ests = [];

for order=orders
    signal_est = run_linear_predictor(signal_with_wgn, samples, order);
    % Ganancia de predicción
    error = signal - signal_est;
    gp = signal'*signal / (error'*error);
    gps = [gps gp];
    signal_ests = [signal_ests, signal_est];
end
```

Plot ganancia de predicción vs orden

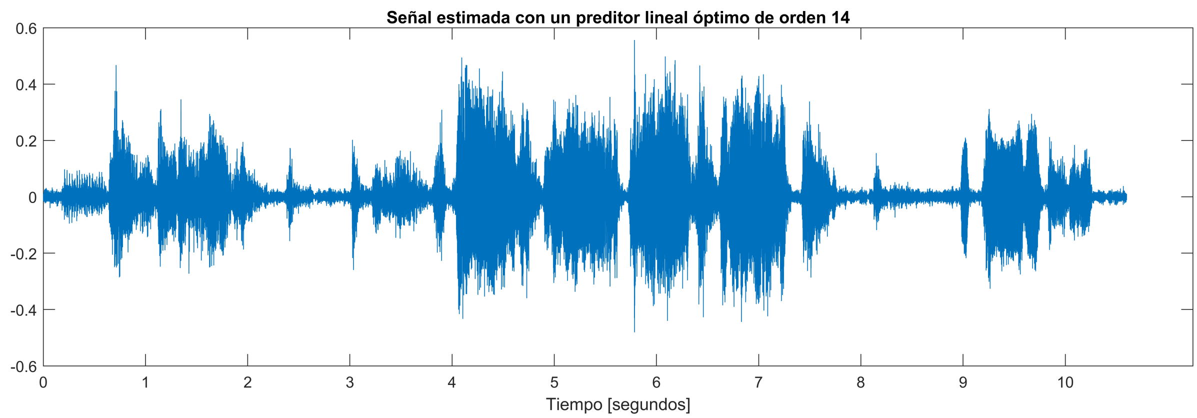
```
bar(orders, gps);
xlabel("Orden del predictor");
```

```
ylabel("Ganancia de predicción");
```



Plot de la estimación con orden 14

```
plot(signal_ests(:,5))
xlabel("Tiempo [segundos]")
xticks([0:8000:8000*10])
xticklabels(["0","1","2","3","4","5","6","7","8","9","10"])
title("Señal estimada con un preditor lineal óptimo de orden 14")
```




```
%sound(signal_ests(:,14), Fs)
```

```
%sound(signal, Fs)
```

```
%sound(signal_with_wgn, Fs)
```