

# **Digital Signal Processing Lab**

## Demo 20 - Guitar

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## Solution

Our program implements the Karplus-Strong plucked-string algorithm in real time using Python and PyAudio.

A circular buffer is initialized with white noise to represent the pluck excitation. Each output sample is generated by reading one value from the buffer, then replacing it with the average of the first two samples multiplied by a damping factor  $K$  to simulate energy loss.

The code streams each sample directly to the audio device, ensuring real-time playback without using block processing or `lfilter`.

Key points:

- Sampling rate:  $F_s=8000$  Hz
- Damping factor  $K \approx 0.998$  controls sustain
- Frequency determined by buffer size `textbfN`
- Circular buffer implementation ensures efficient real-time behavior

This setup synthesizes a natural-sounding plucked string tone entirely in Python using direct sample-by-sample processing.

## Addendum

Full Code

```
1 import numpy as np
2 import pyaudio
3 import struct
4
5 def karplus_strong(
6     frequency=150.0,
7     Fs=8000,
8     K=0.998):
9
10    duration=10*50/frequency
11    #print(duration)
12    print("* Playing")
13    N = int(Fs / frequency)
14    buffer = np.random.uniform(-1, 1, N)
15    p = pyaudio.PyAudio()
16    stream = p.open(format=pyaudio.paInt16, channels=1, rate=Fs, output=True)
17
18    num_samples = int(duration * Fs)
19    MAXVALUE = 2**15 - 1
20
21    # circular buffer index
22    idx = 0
23
24    for i in range(num_samples):
25        y = buffer[idx]
26        next_idx = (idx + 1) % N
27        avg = 0.5 * (buffer[idx] + buffer[next_idx])
28        buffer[idx] = K * avg # feedback
29        idx = next_idx
30        # convert to bytes and play
31        sample = struct.pack('h', int(y * MAXVALUE))
32        stream.write(sample)
33
34    stream.stop_stream()
35    stream.close()
36    p.terminate()
37    print("* Finished")
38
39 karplus_strong(110,K=0.998)
```

Snippet 1: Complete implementation