# Sampling and Aliasing

Eddie Franco[1], Kyle G. Gayliyev[1], Skylar Stockham[1]

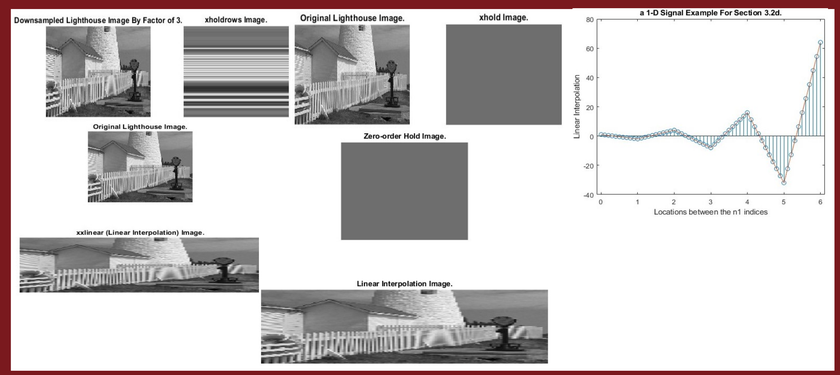
[1] Department of Electrical and Copmputer Engineering; University of Utah; Salt Lake city, UT

## Background

When sampling to convert a continuous-time (or analog) signal to a digital form for computer processing and storage, the primary issue is aliasing and the sampling strategy necessary to avoid aliasing of frequency components.

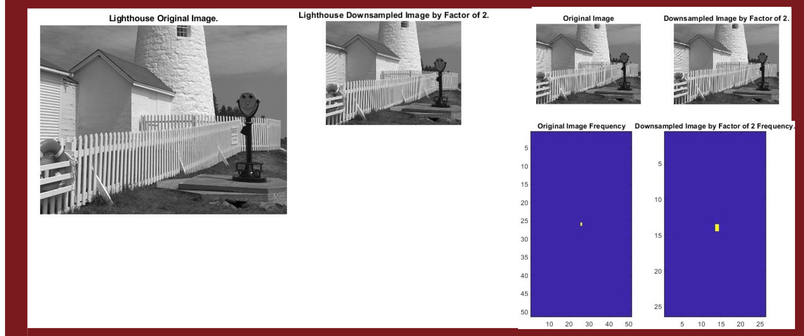
The objective of our presentation is to understand the Sampling Theorem which states that the sampling rate must be greater than twice the highest frequency contained in the analog signal. Frequency content is taken to mean the spectral content of a signal when represented as a sum of sinusoids.

We present the signal reconstruction of a D-to-A converter from a practical point of view as a generalization of interpolation.

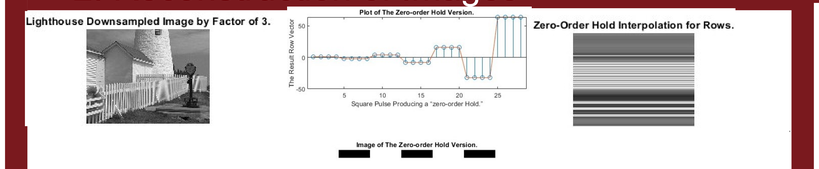



## Lab P-8: Digital Images: A/D and D/A

### 1. Down-Sampling




### 2. Reconstruction of Images




## Lab S-8: Spectrograms: Harmonic Lines & Chirp Aliasing

time (ms)

## Summary

## References

1. James H. McClellan, Ronald W. Schafer. 4. Sampling and Aliasing, dspfirst.gatech.edu/chapters/04samplin/overview.html. Accessed 2 Dec. 2024.
2. Proakis, John G., and Dimitris G. Manolakis. Digital Signal Processing. Prentice Hall, 2006.