What is Signal Processing?

Signal Processing

Digital vs Analog

Digital **S**ignal **P**rocessing (DSP)

Sampling frequency (Rate)

Aliasing

Nyquist's Theorum

Quantization and Quantization Noise

Peak clipping and Dynamic Range

Overflow (Wrapping)

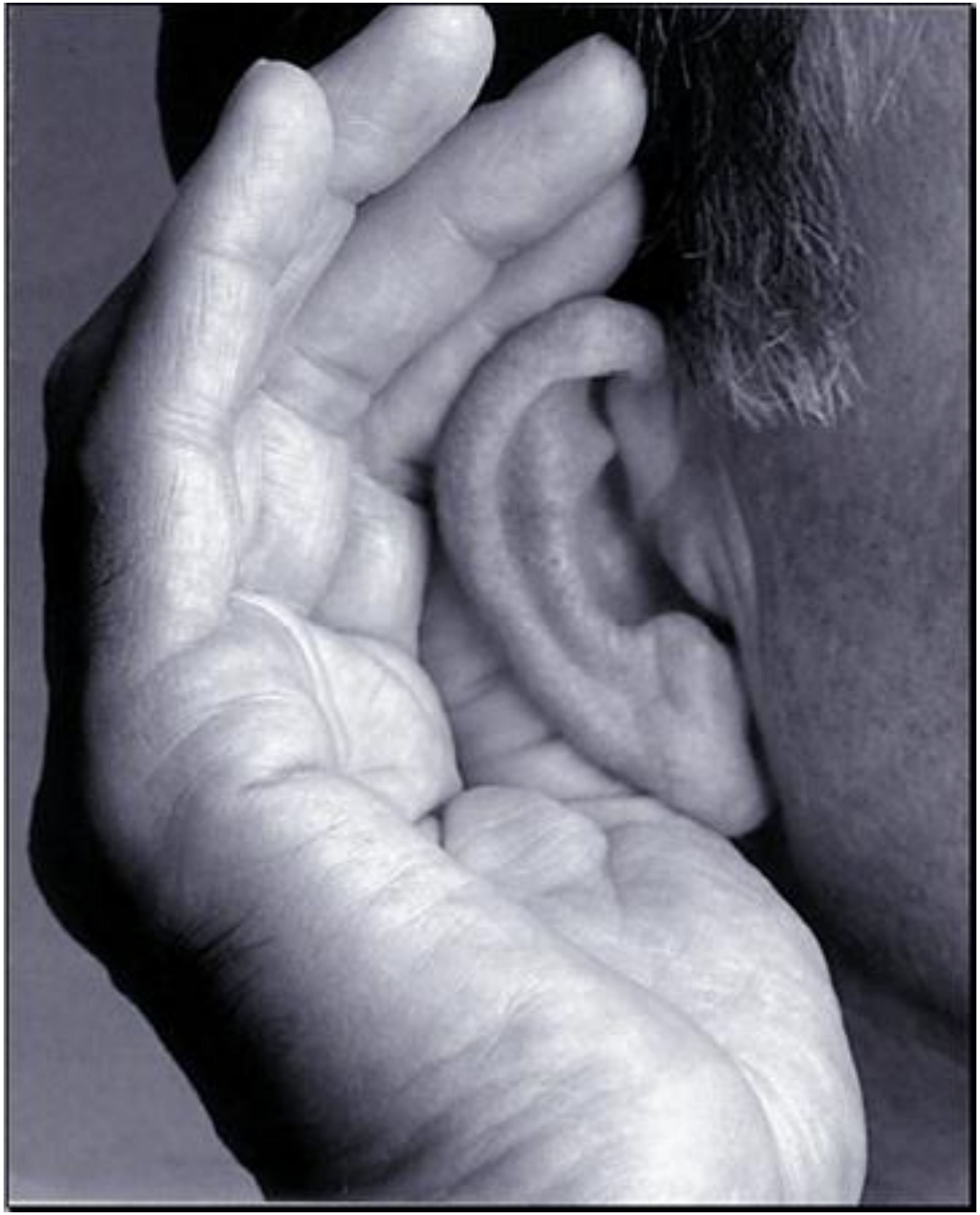
Extraction,

enhancement,

or recovery

of information from a signal.

We have
built-in signal processing

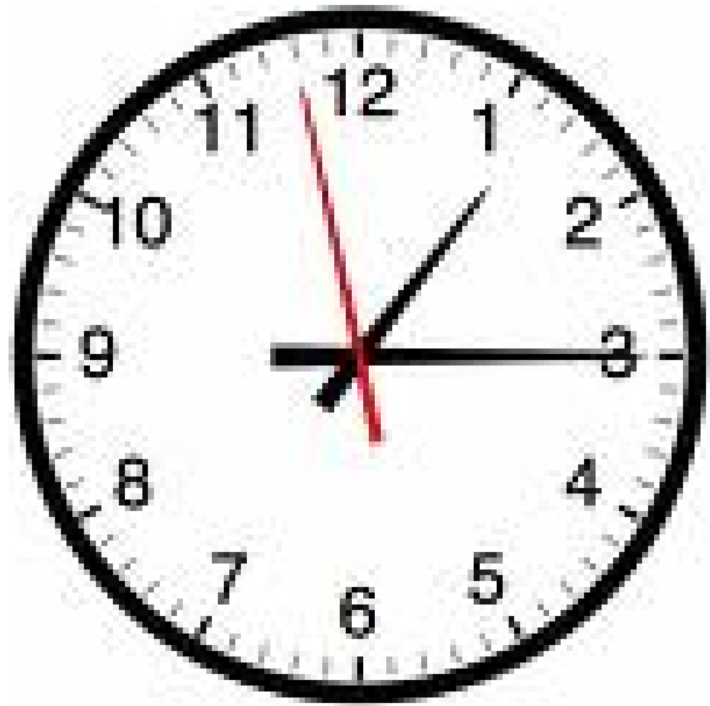


And

We have instruments to enhance signal processing.


CARSON


35
36
37
38
39
40
41
42


12
1
2
3
4
5
6
7
8
9
10
11

Natural signals vary along a continuum.

Such natural continuous signals are **analog.**

BUT

the most amazing advances in signal processing have been made possible by computers.

To be analyzed with computers…

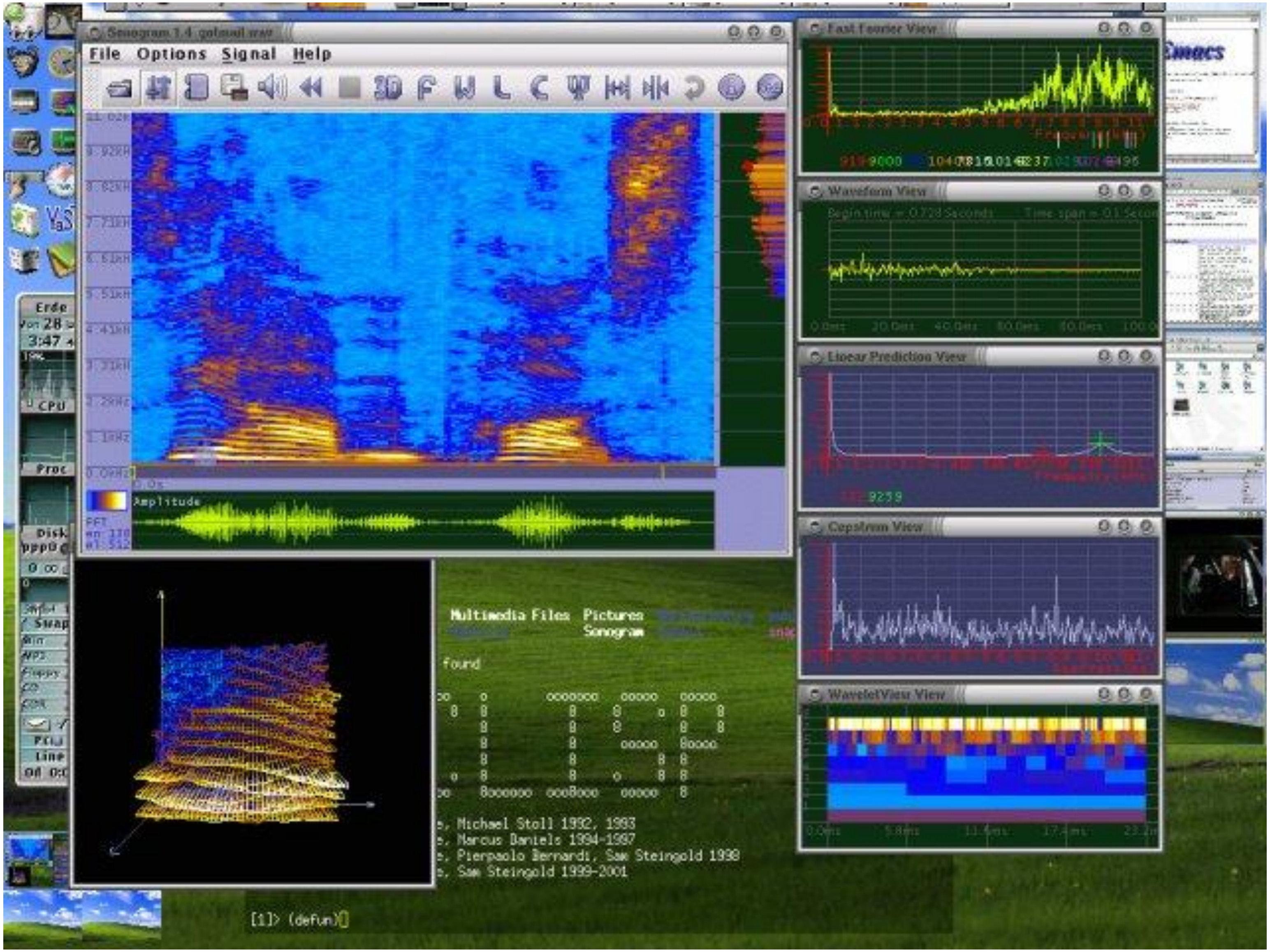
Signals must be digitized.

So, the whole field is called

Digital
Signal
Processing

and it’s Ubiquitous

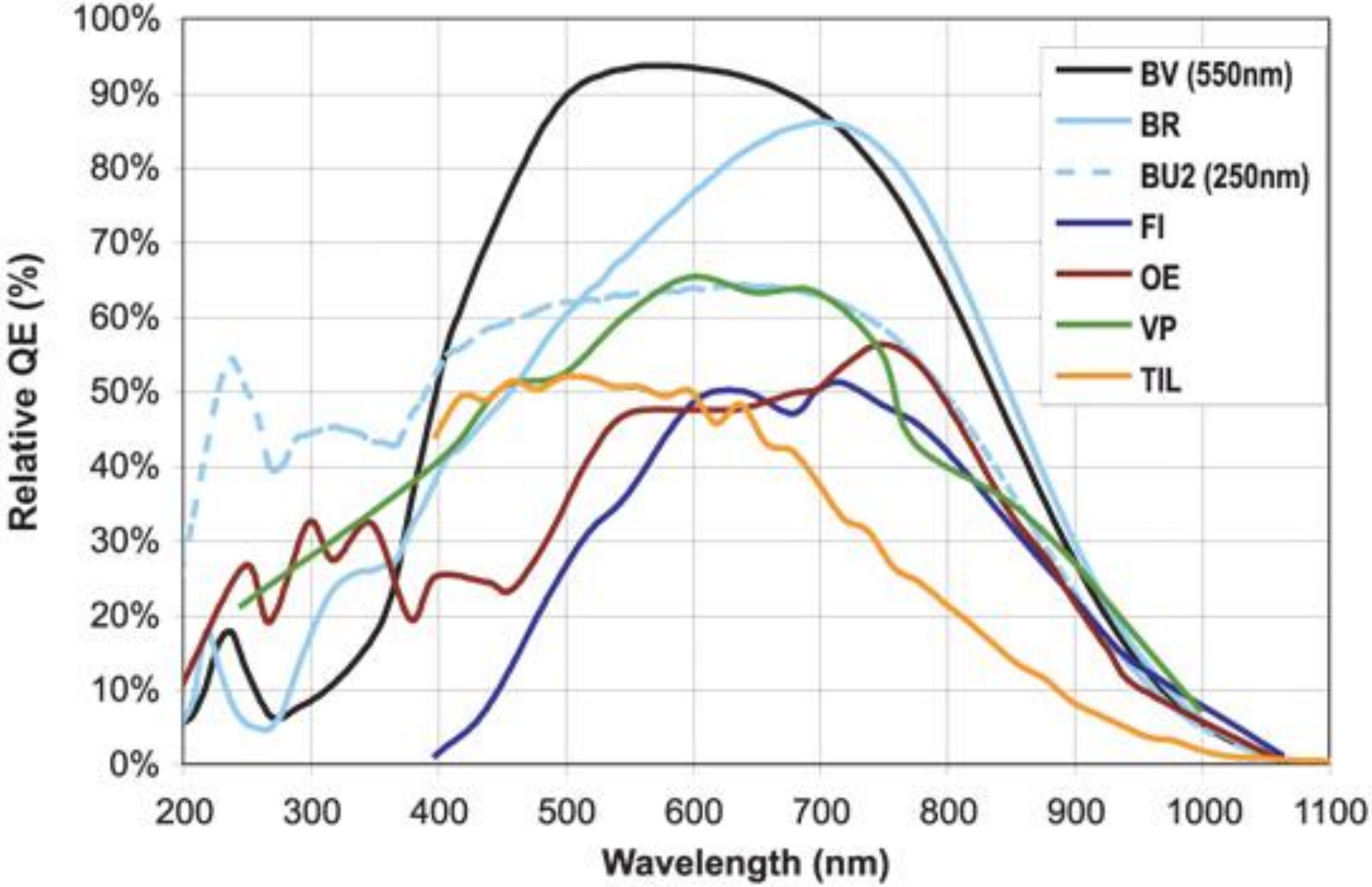
Acoustic Analysis



Art

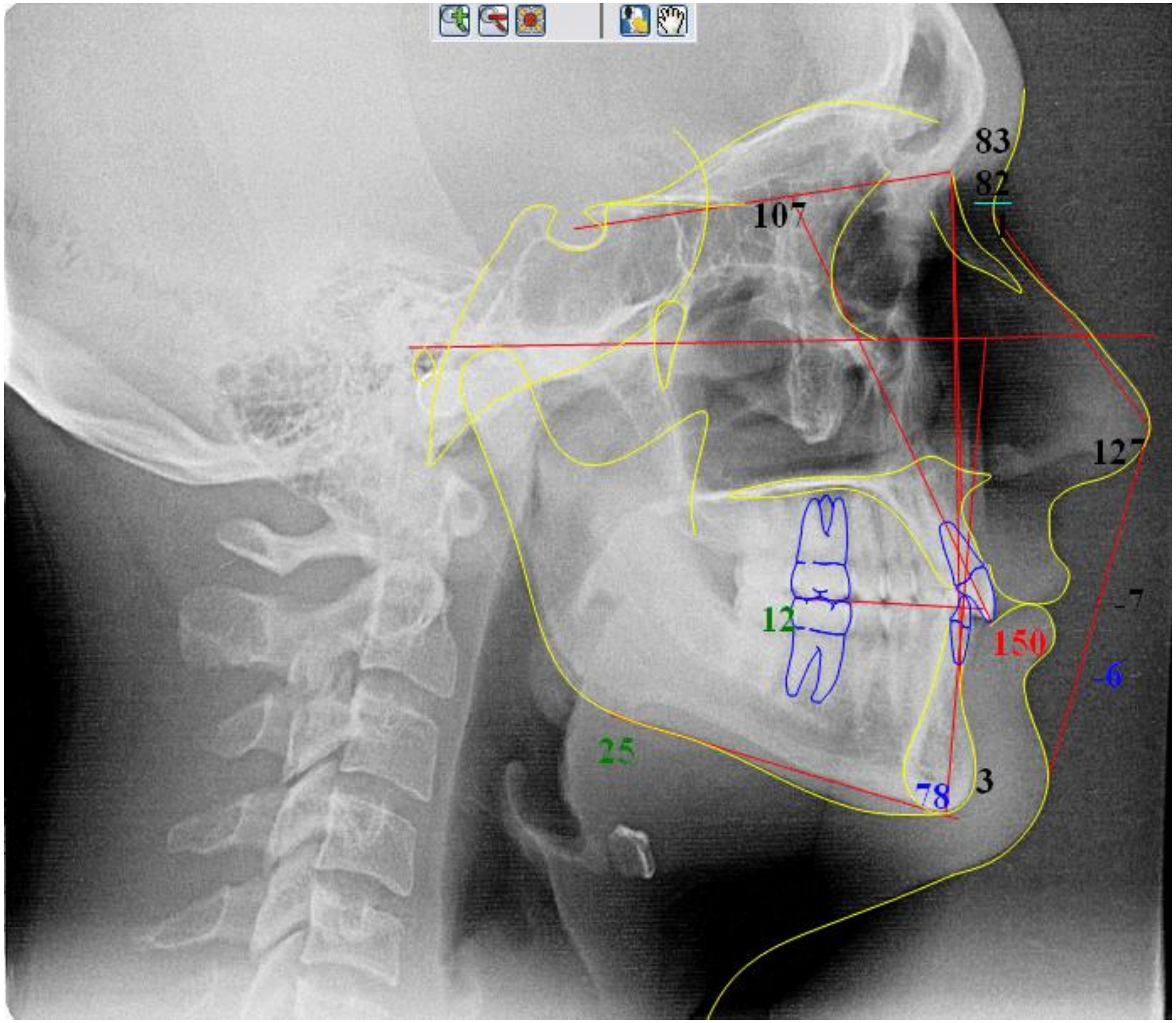

Canon
EOS
CANON ZOOM LENS EF-S 18-55mm 1:3.5-5.6 II
CANON INC.
DIGITAL
REBEL
XTi

Astronomy

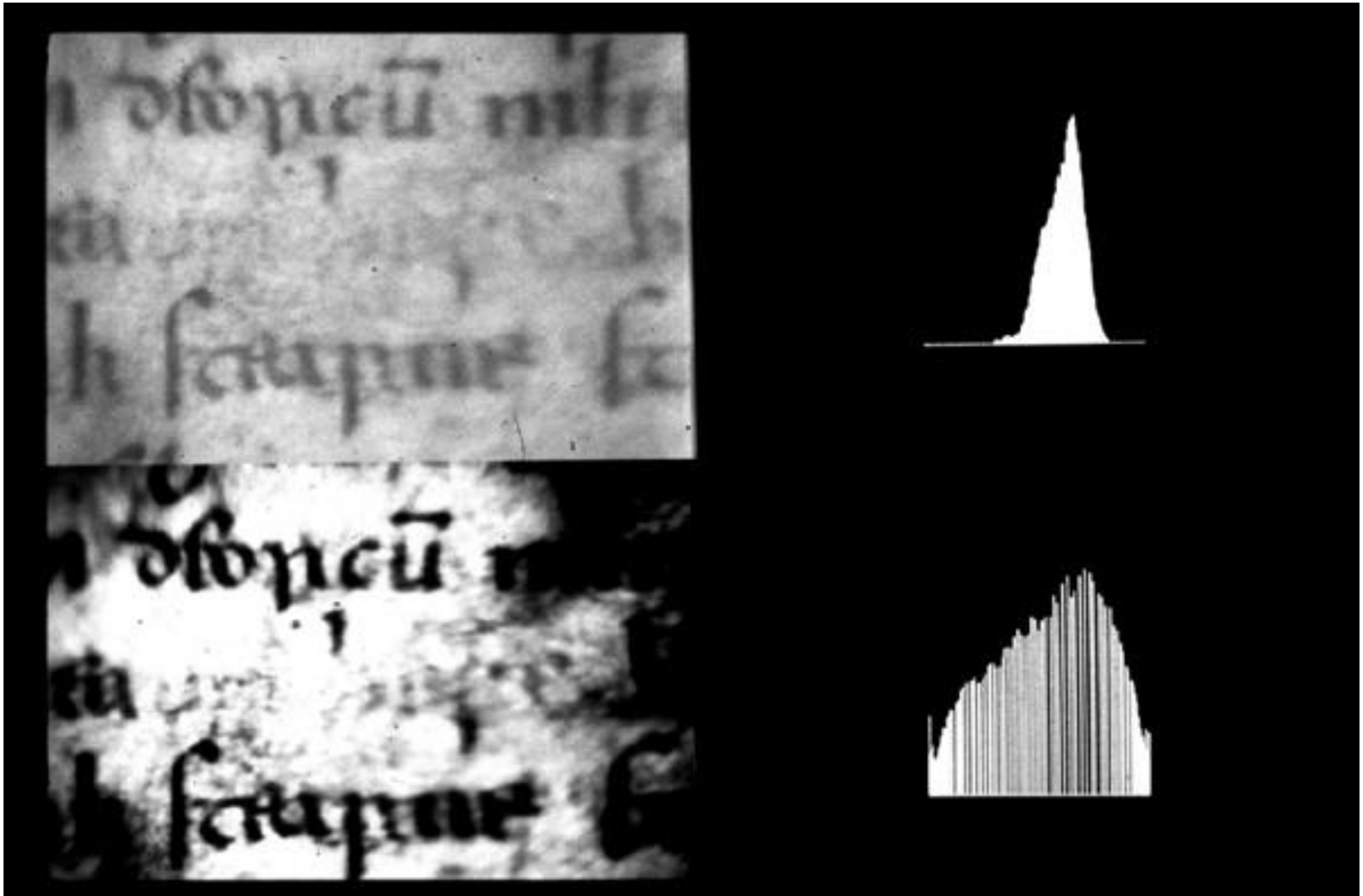


100%
90%
80%
70%
60%
50%
40%
30%
20%
10%
0%
Relative QE (%)
200
300
400
500
600
700
800
900
1000
1100
Wavelength (nm)
BV (550nm)
BR
BU2 (250nm)
FI
OE
VP
TIL

Dentistry



Document Restoration



[…]ende þa gyt hie him asetton segen g 130

denne heah ofer heafod leton holm ber

geafon on gar secg him wæs geomor sefa

murnende mod men ne cunnon . secgan to

soðe sele rædenne hæleð under heofen

hwa þæm hlæste on feng.

.I.

ÐA wæs on burgum beowulf scyldinga le

leod cyning longe þrage folcum gefræ

ge fæder ellor hwearf aldor of earde

oþ ꝥ him eft onwoc heah healf dene heold

þenden lifde gamol ⁊ guð reouw glæde scyl

dingas ðæm feower bearn forð gerimed in

worold wocun weoroda ræswa heorogar ⁊

hroðgar ⁊ halga til hyrde ic ꝥ elan cwen

heaðo scilfingas heals gebedda þa wæs hroð

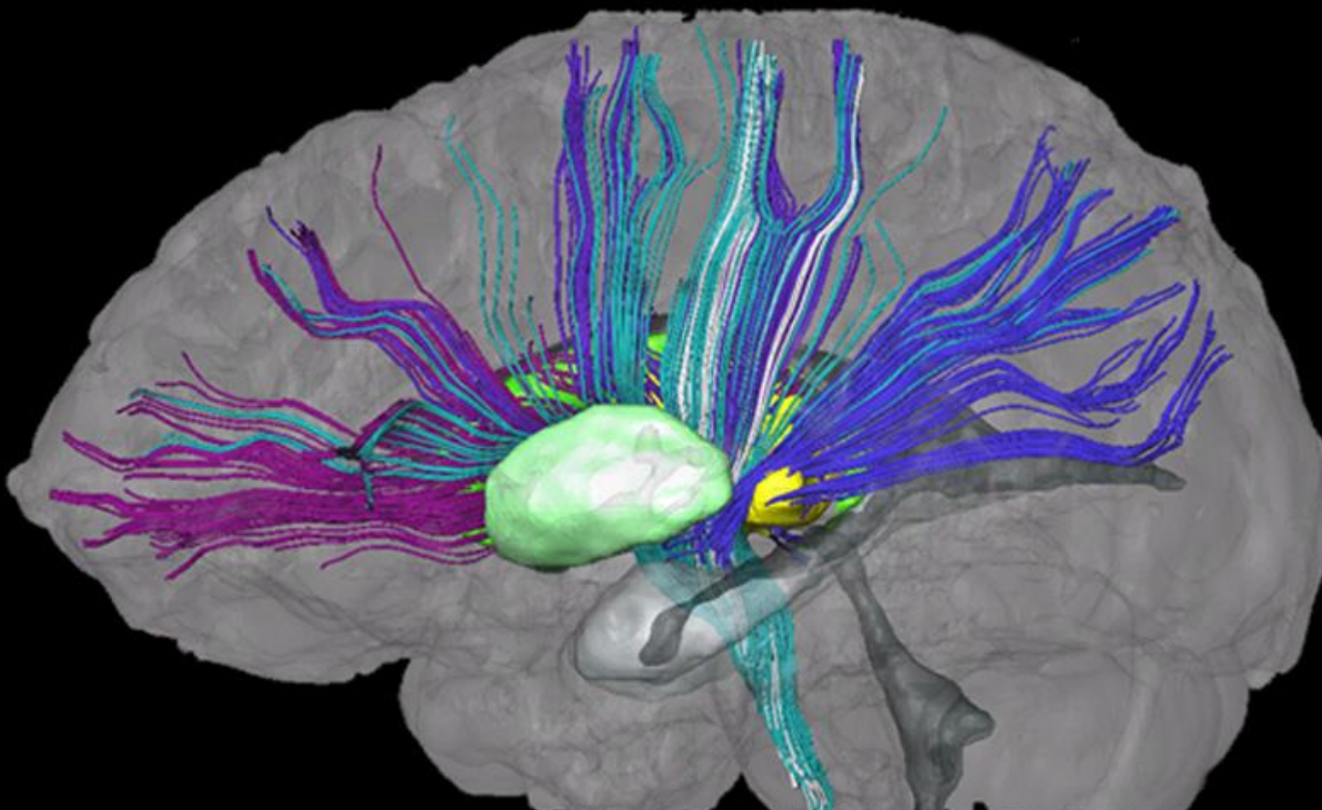
gare here sped gyfen wiges weorð mynd ꝥ

him his wine magas georne hyrdon oðð ꝥ

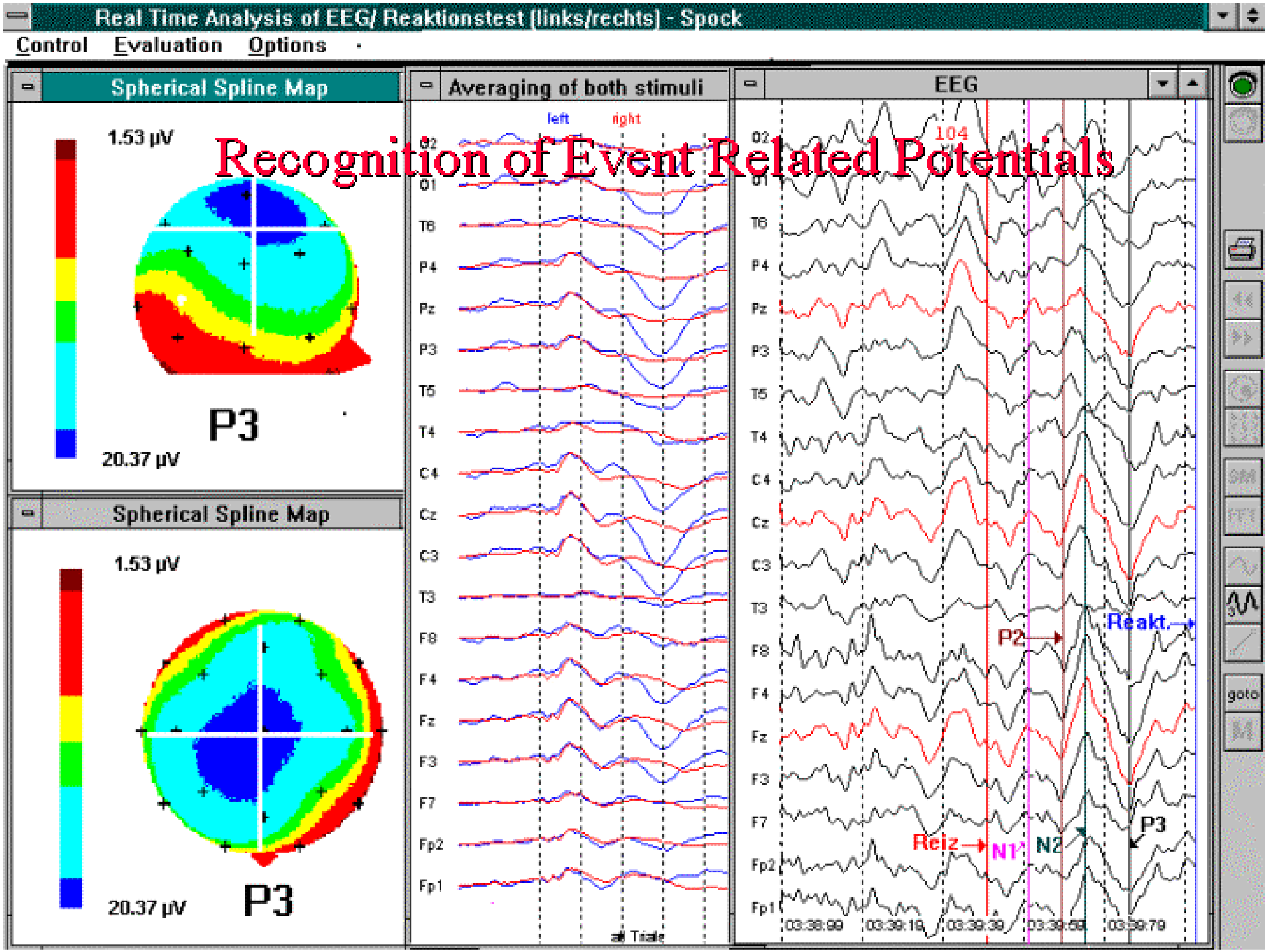
seo geo goð geweox mago driht micel him

on mod bearn ꝥ heal reced hatan wolde

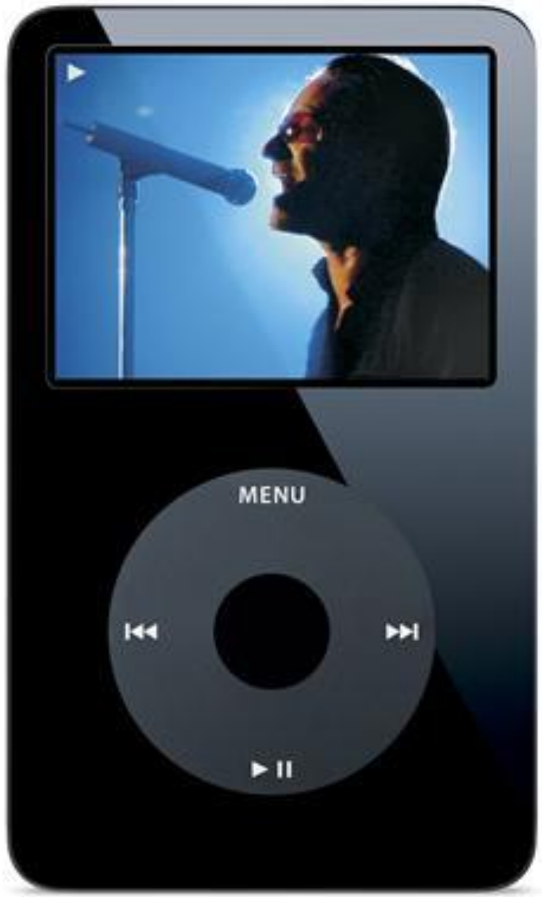
DTI



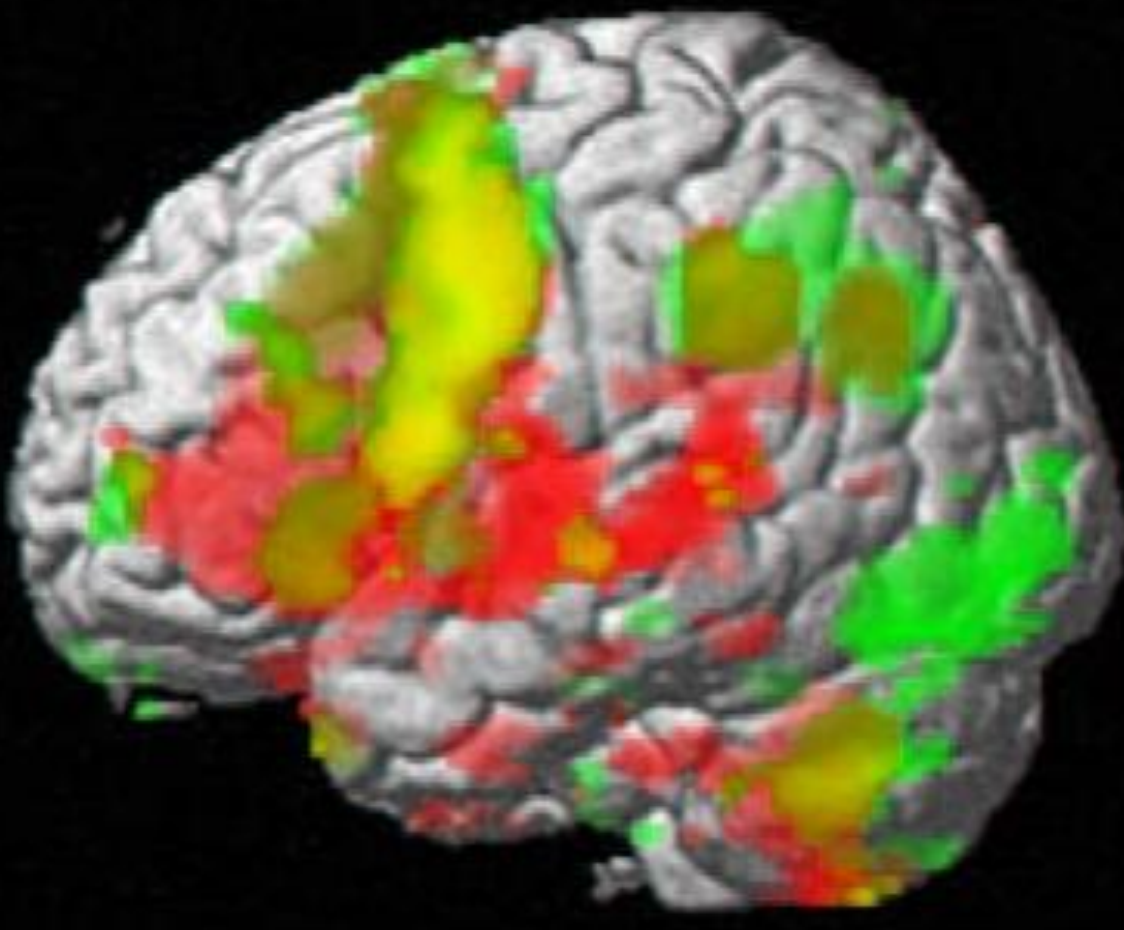
EEG


Real Time Analysis of EEG/ Reaktionstest (links/rechts) - Spock
Control Evaluation Options
Spherical Spline Map
1.53 µV
20.37 µV
P3
Spherical Spline Map
1.53 µV
20.37 µV
P3
Averaging of both stimuli
left
right
all Trials
EEG
104
Recognition of Event Related Potentials
Reiz
N1
N2
P2
P3
Reakt.
goto

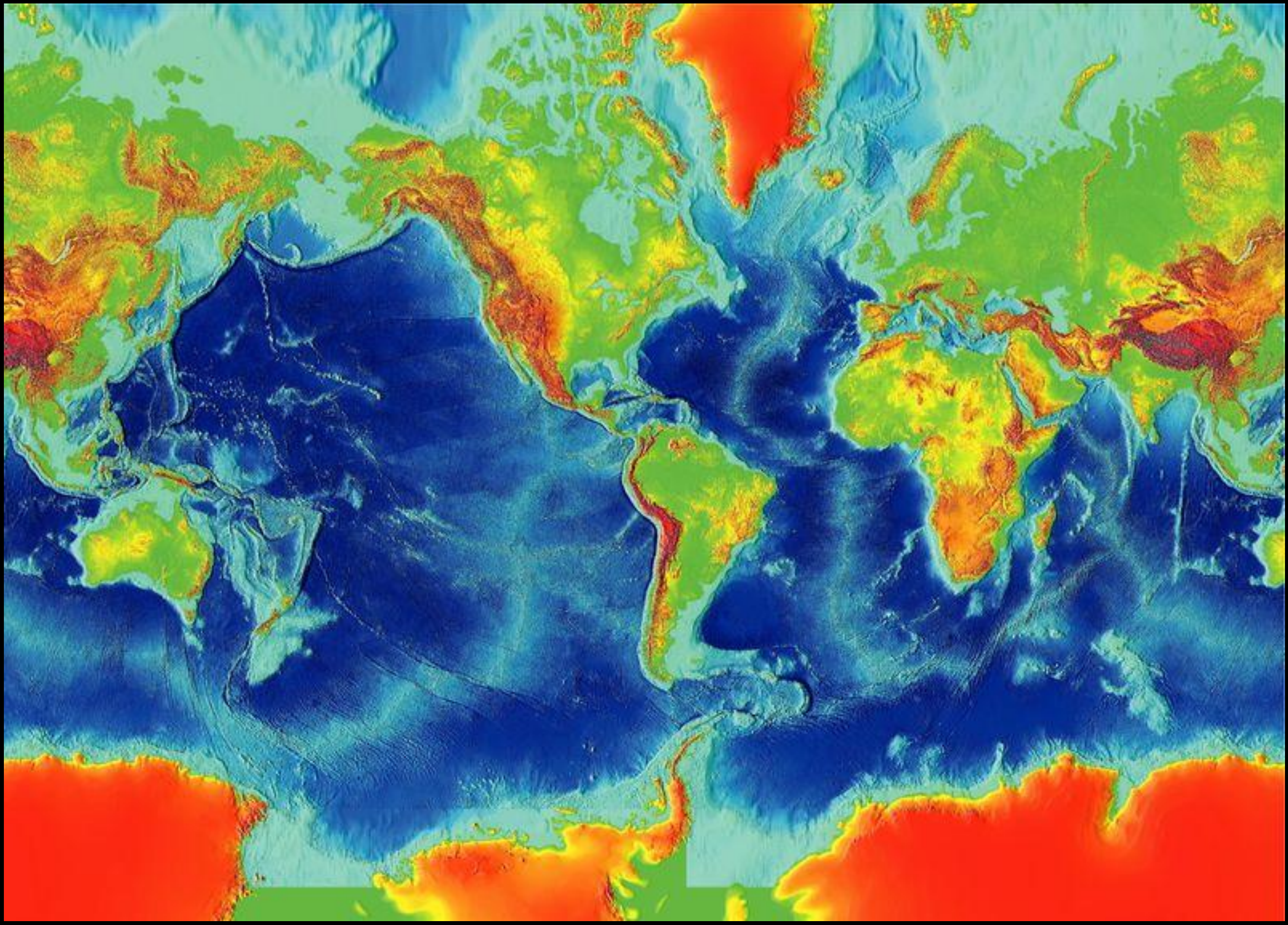
Entertainment



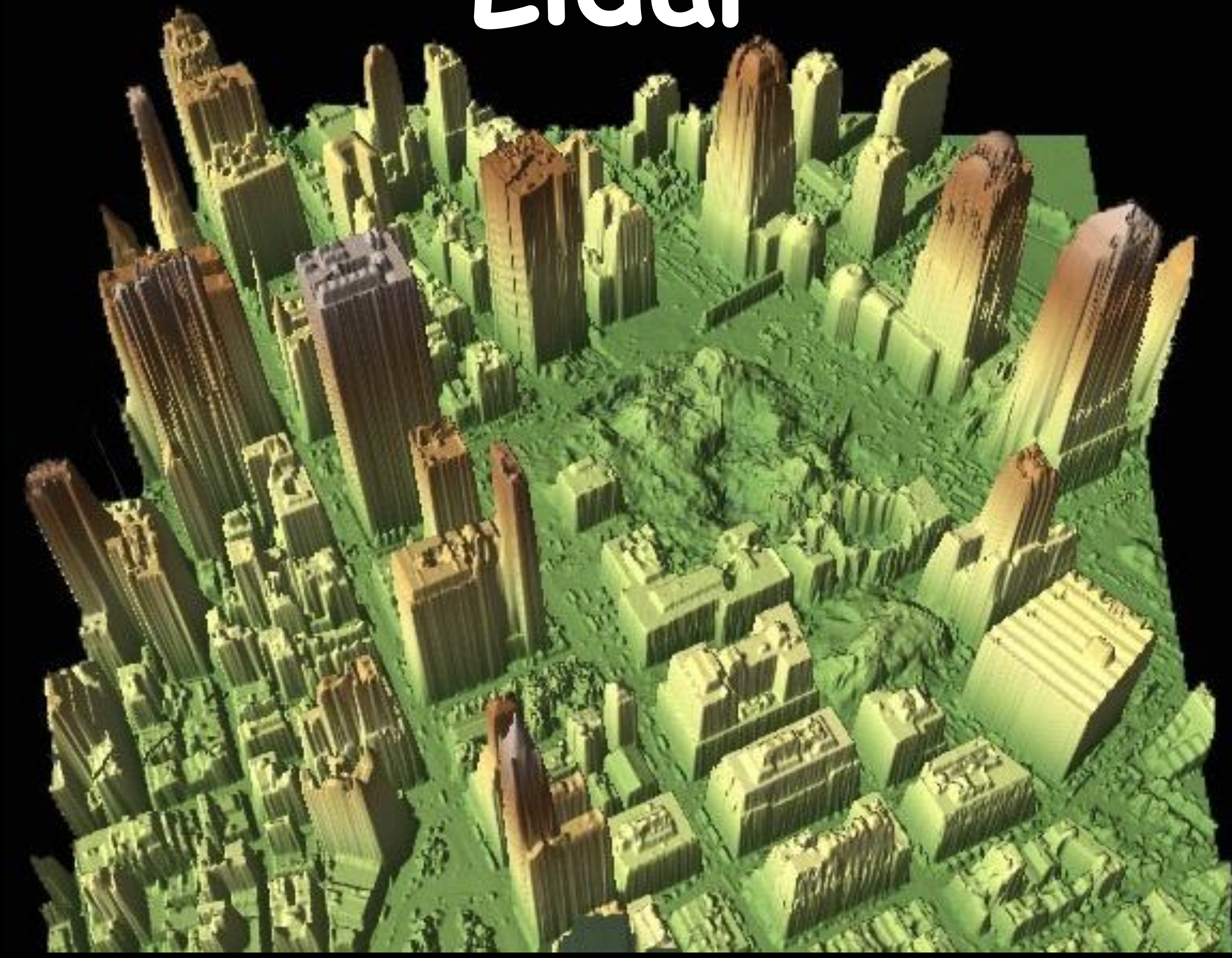
fMRI



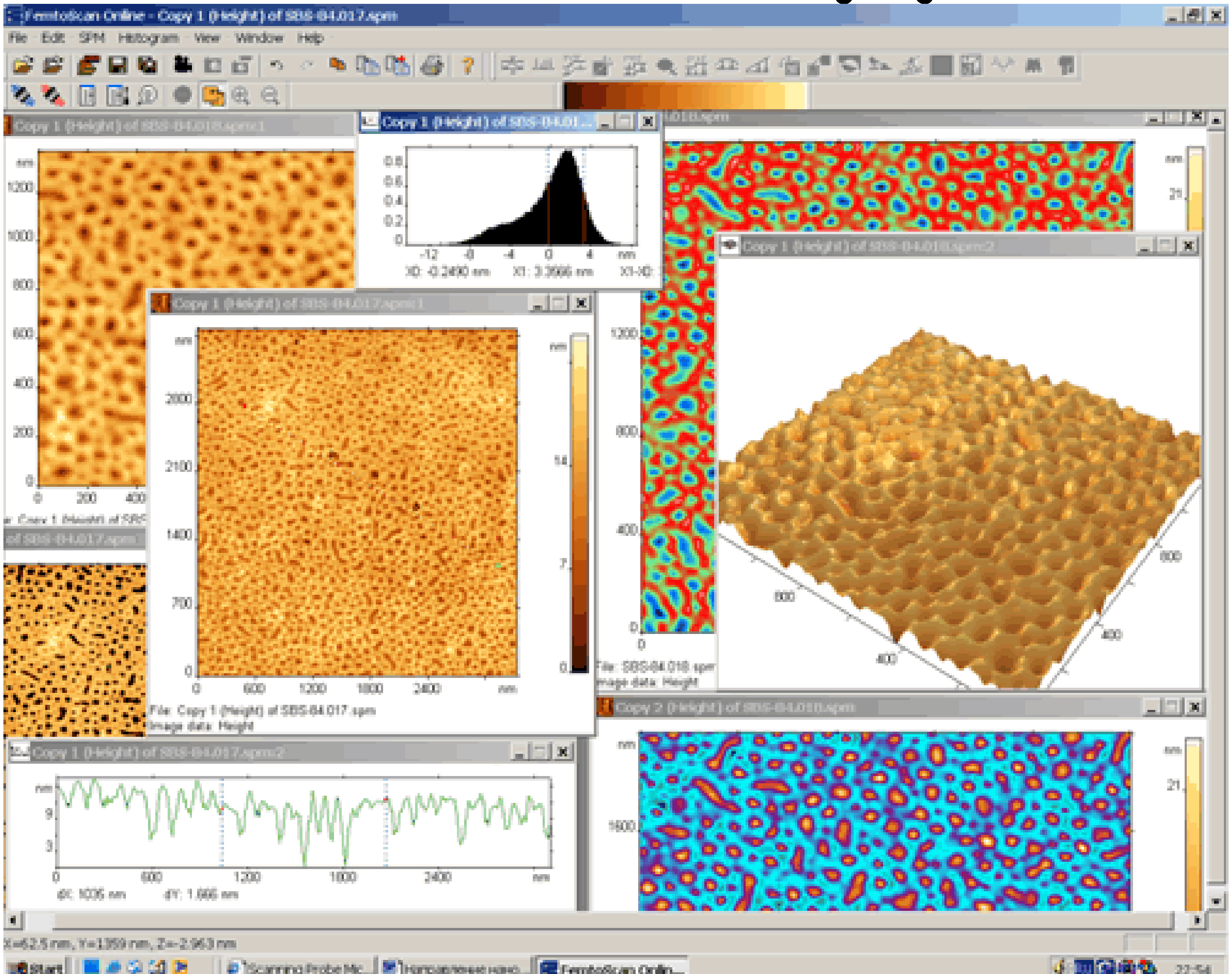
Geo-Imaging



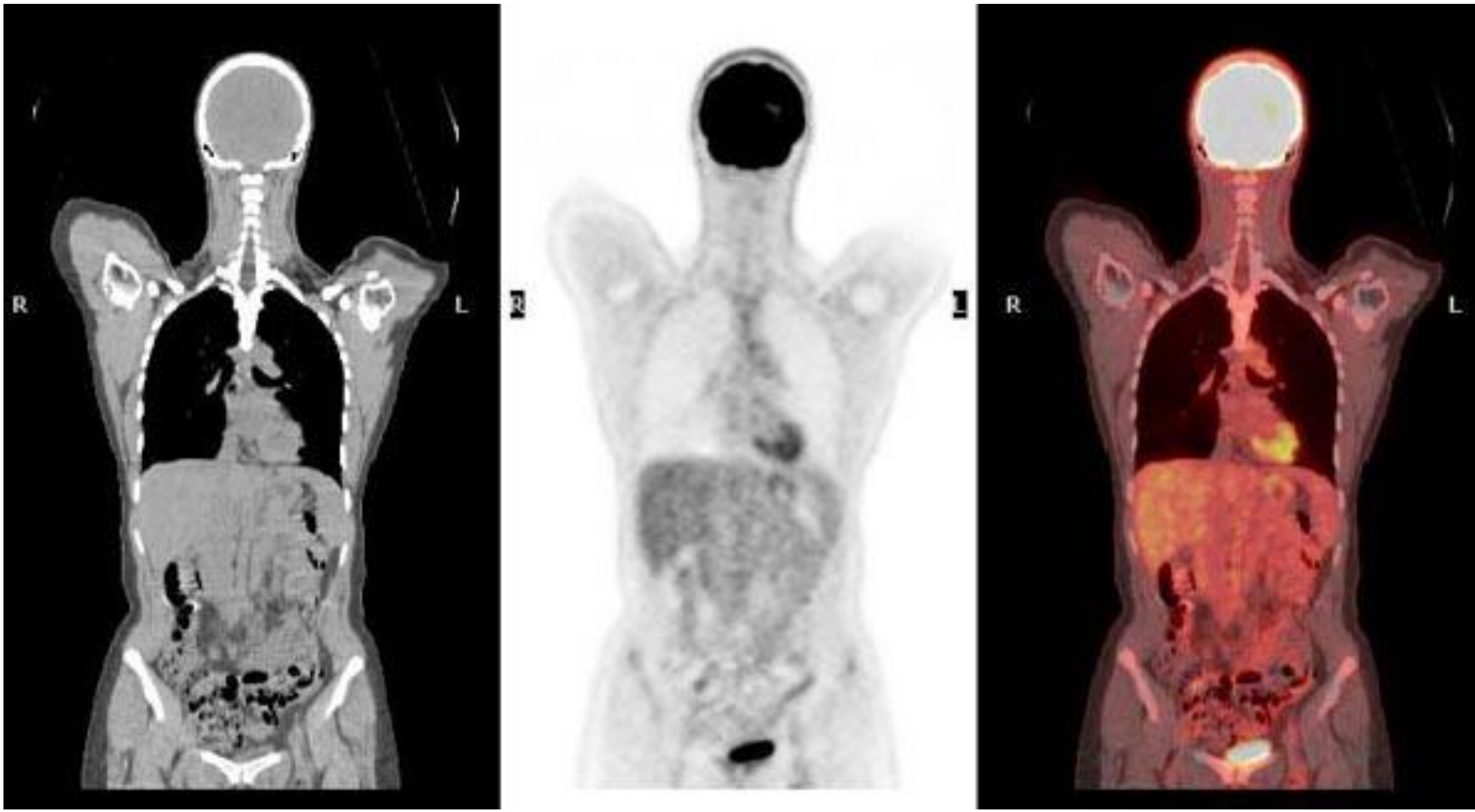
Lidar



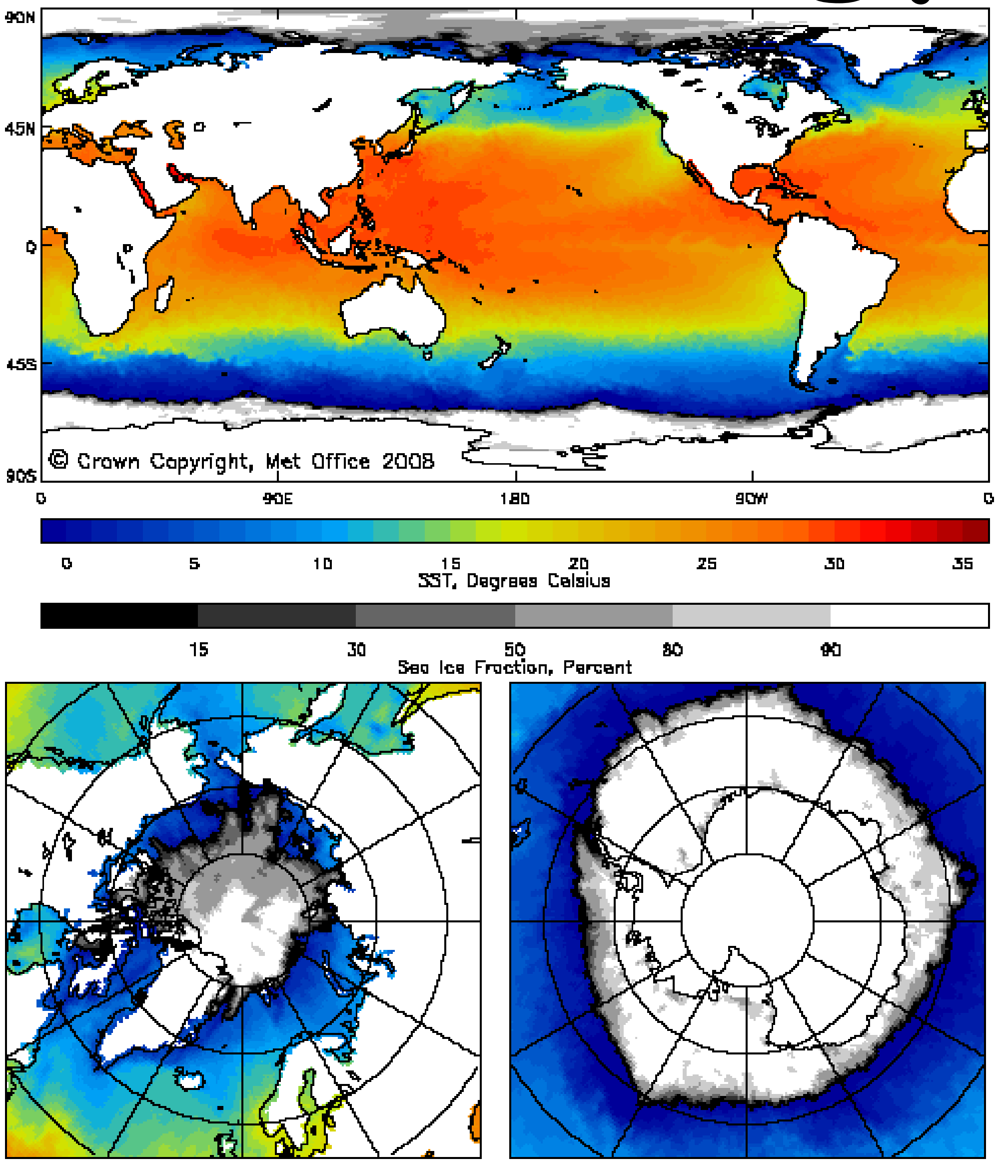
Microscopy



PET Scanning

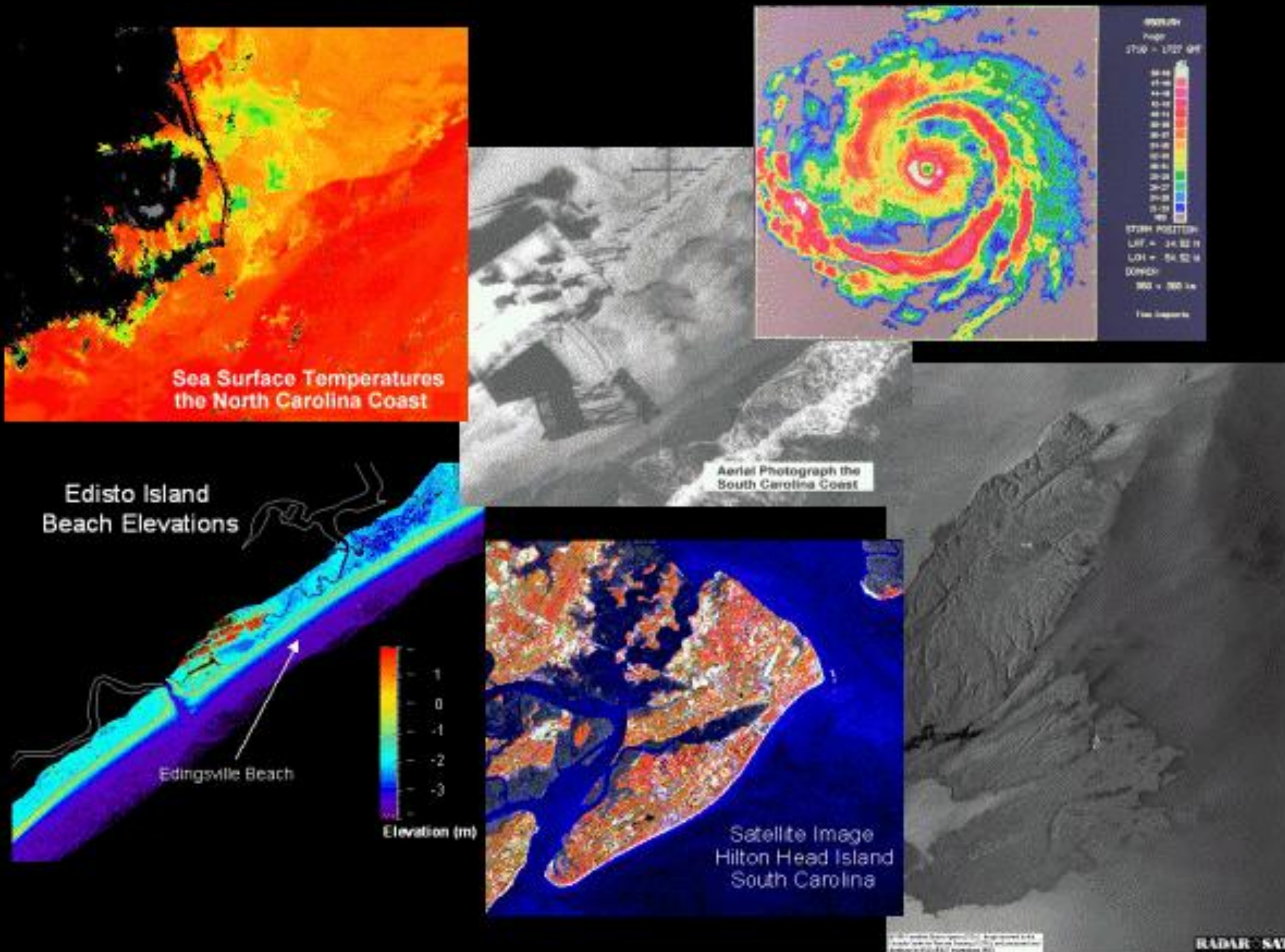


Meteorology



Regional 3D Radar
08/16/08 10:36 PM CT
KSN
WeatherLab
RAIN
MIXED
SNOW
LIGHT MEDIUM HEAVY

Remote Sensing



Seismology

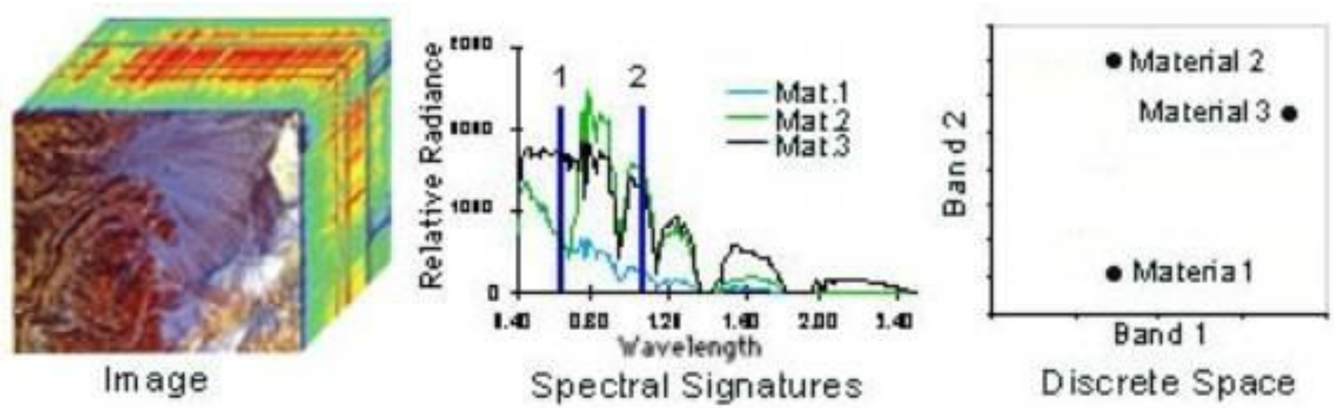
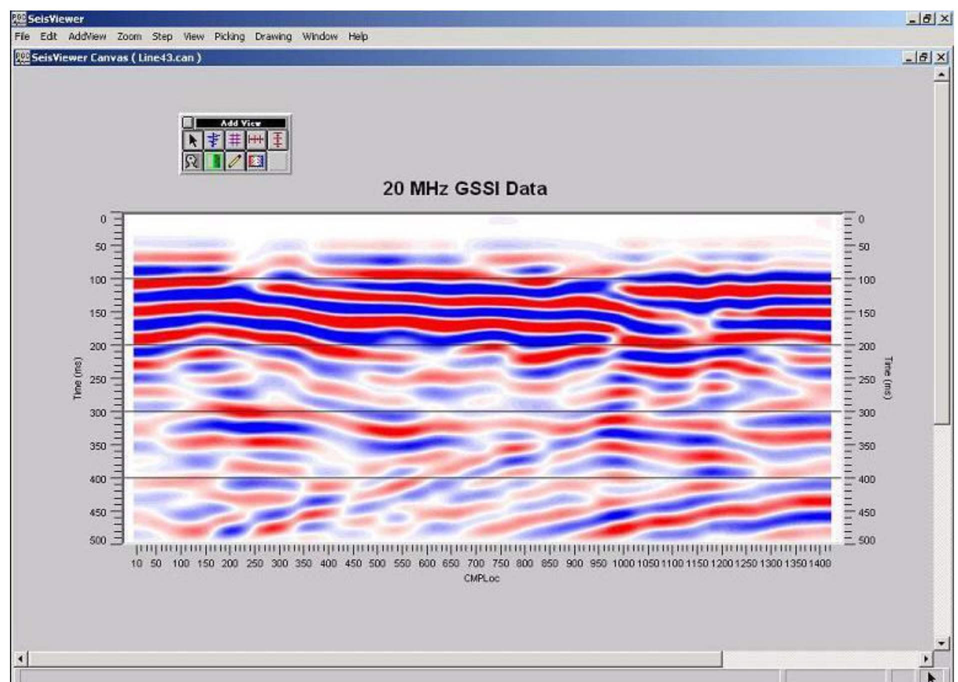
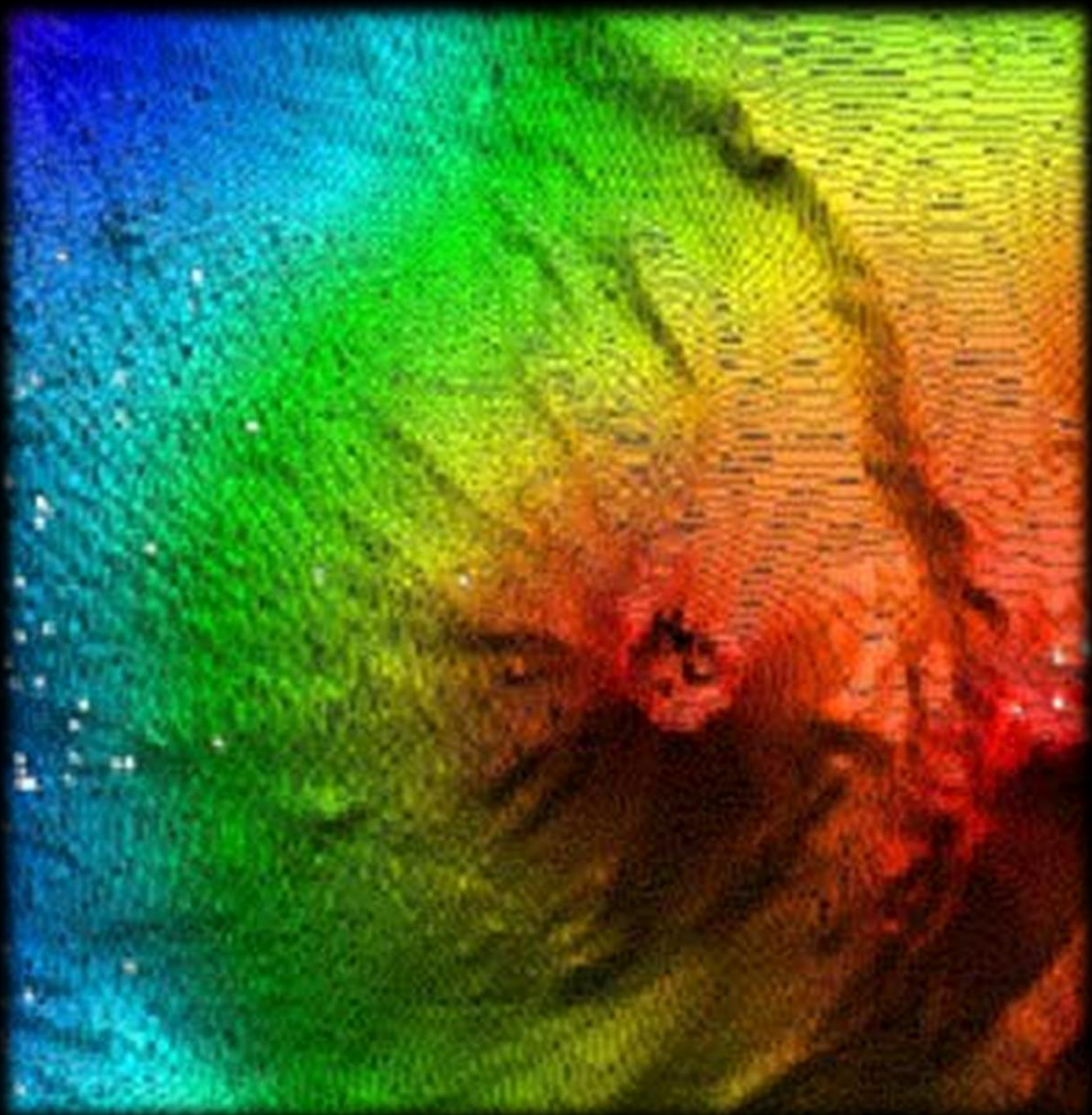


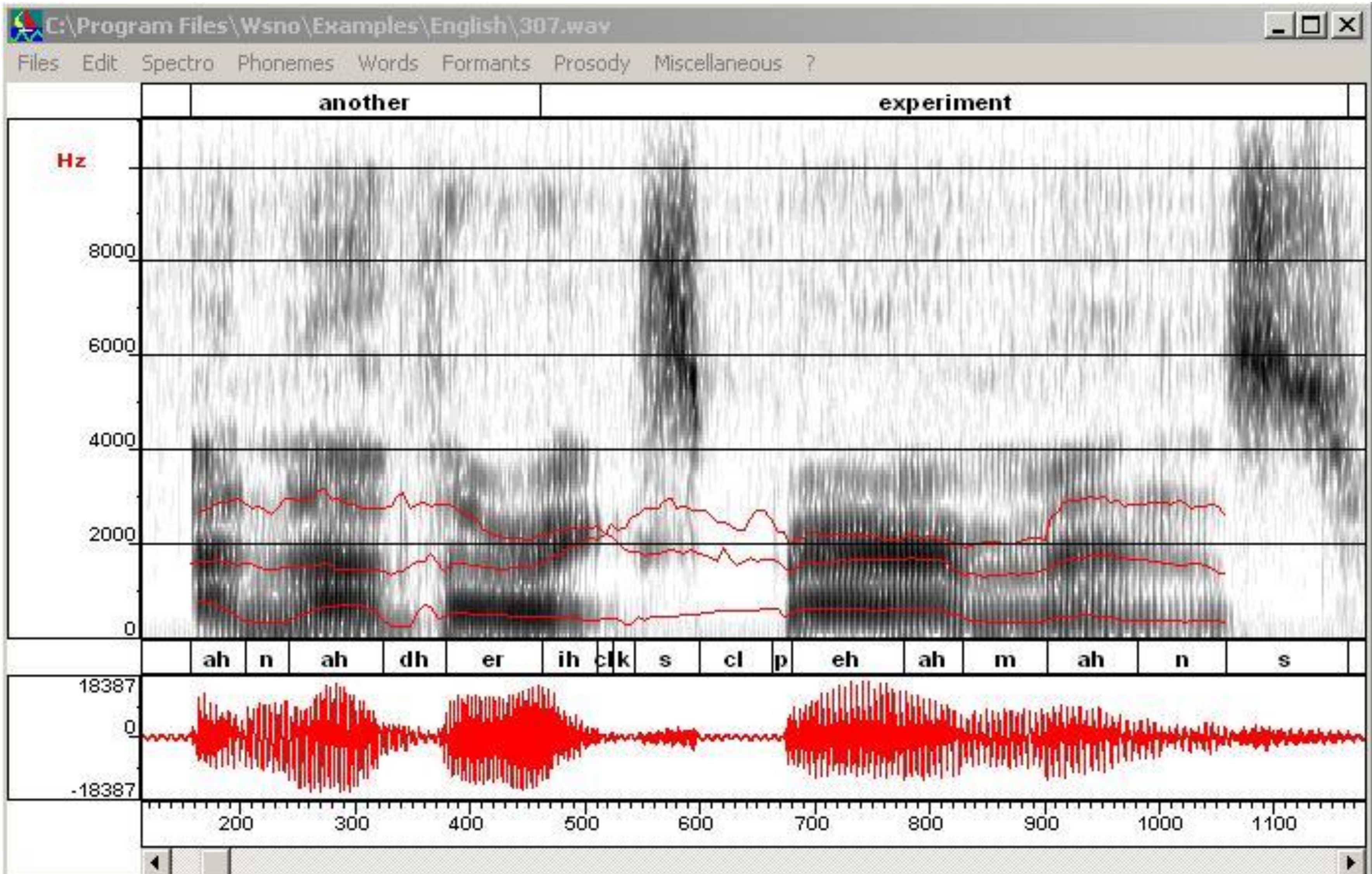
Image
Relative Radiance
1
2
Mat.1
Mat.2
Mat.3
Wavelength
Spectral Signatures
Band 2
Material 2
Material 3
Material 1
Band 1
Discrete Space

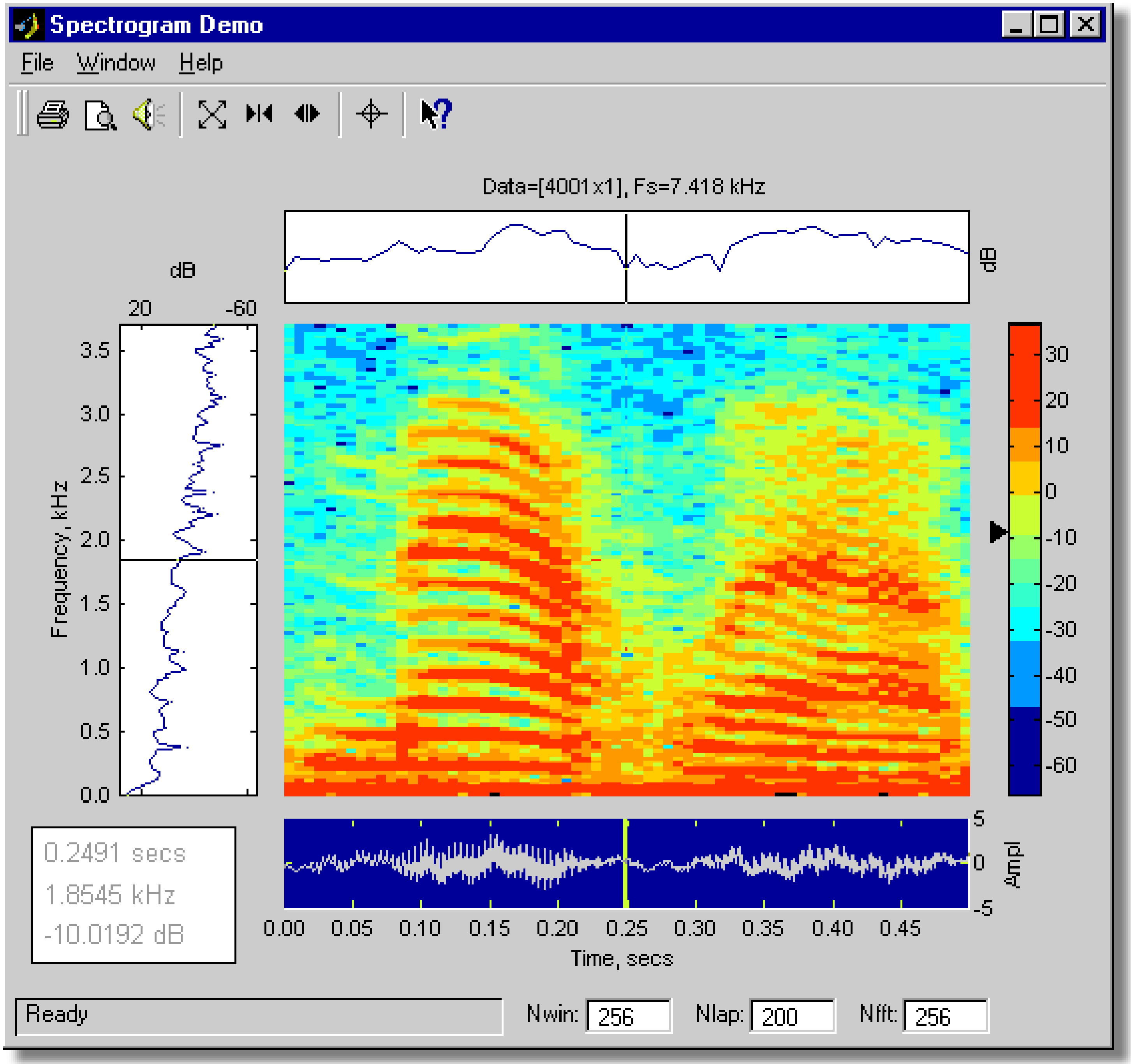

SeisViewer
File Edit AddView Zoom Step View Picking Drawing Window Help
SeisViewer Canvas (Line43.can)
Add View
20 MHz GSSI Data
Time (ms)
CMPLoc

Sonar



and Voice Analysis




Spectrogram Demo
File
Window
Help
Data=[4001x1], Fs=7.418 kHz
dB
dB
20
-60
Frequency, kHz
3.5
3.0
2.5
2.0
1.5
1.0
0.5
0.0
30
20
10
0
-10
-20
-30
-40
-50
-60
5
0
-5
Ampl
0.2491 secs
1.8545 kHz
-10.0192 dB
0.00
0.05
0.10
0.15
0.20
0.25
0.30
0.35
0.40
0.45
Time, secs
Ready
Nwin:
256
Nlap:
200
Nfft:
256

to name a few.

So, what is **digitization**?

Representing an otherwise continuous signal with a **discrete set of samples.**

Digital
Discrete

Analog
Continuous

How many samples are there in that "discrete set of samples"?

Good Question

Sampling Frequency (Sampling Rate) is the number of samples we take…

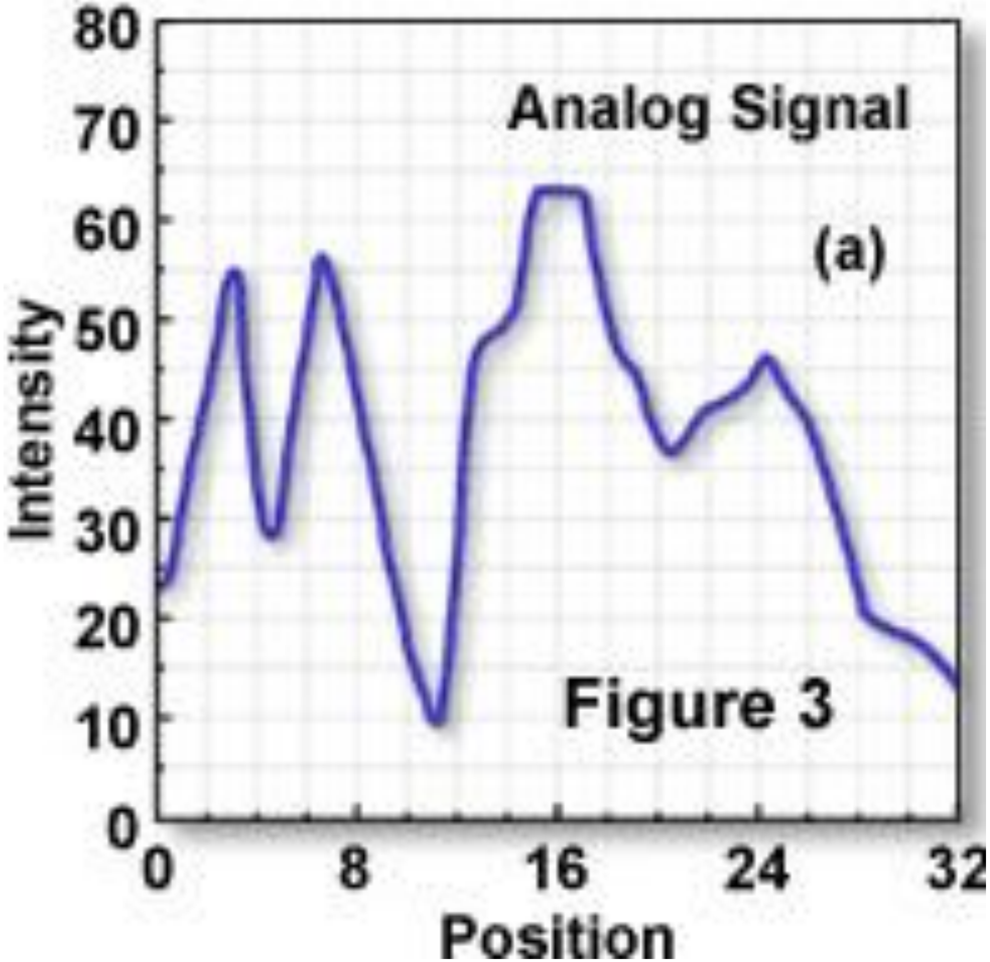
per unit of time (e.g., per second for sounds)

Or per unit of space (e.g., per picture height)

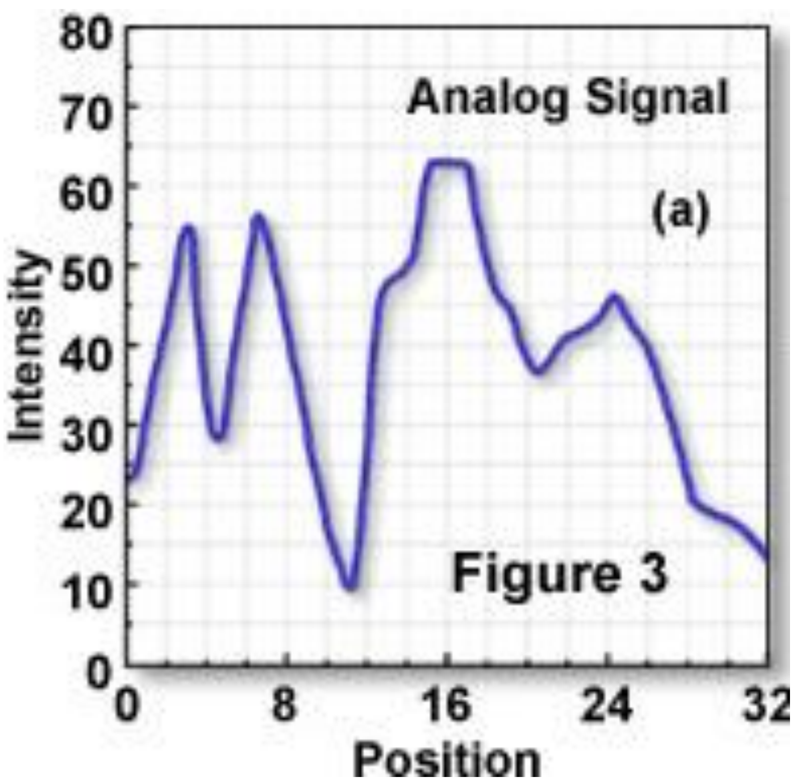
The higher the sampling frequency,the better the resolution,

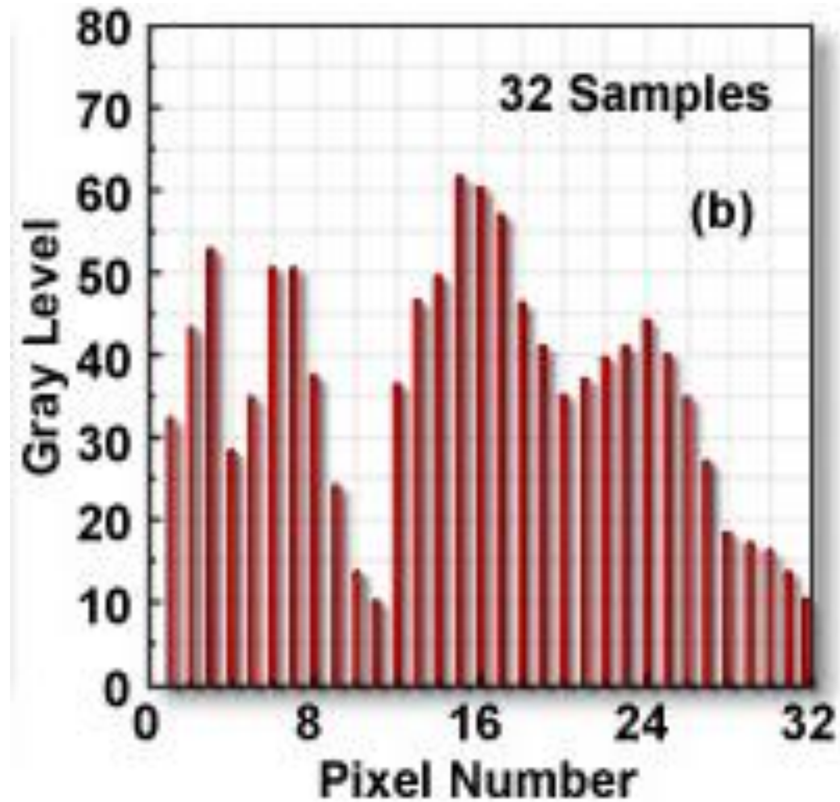
but the better the resolution, the more space it takes to store on your hard drive.

Here's a continuous analog signal

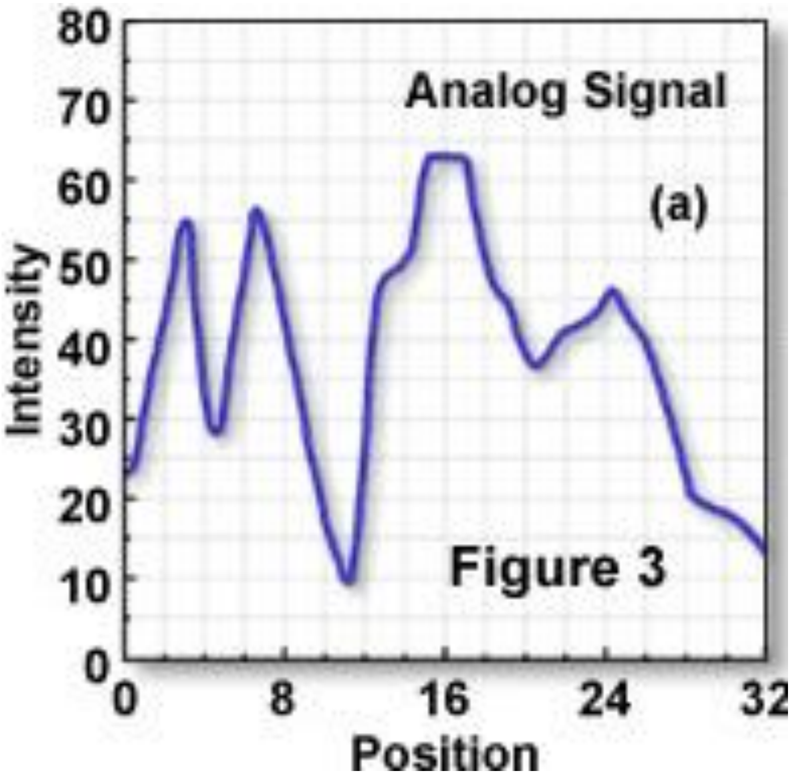


We can sample frequently

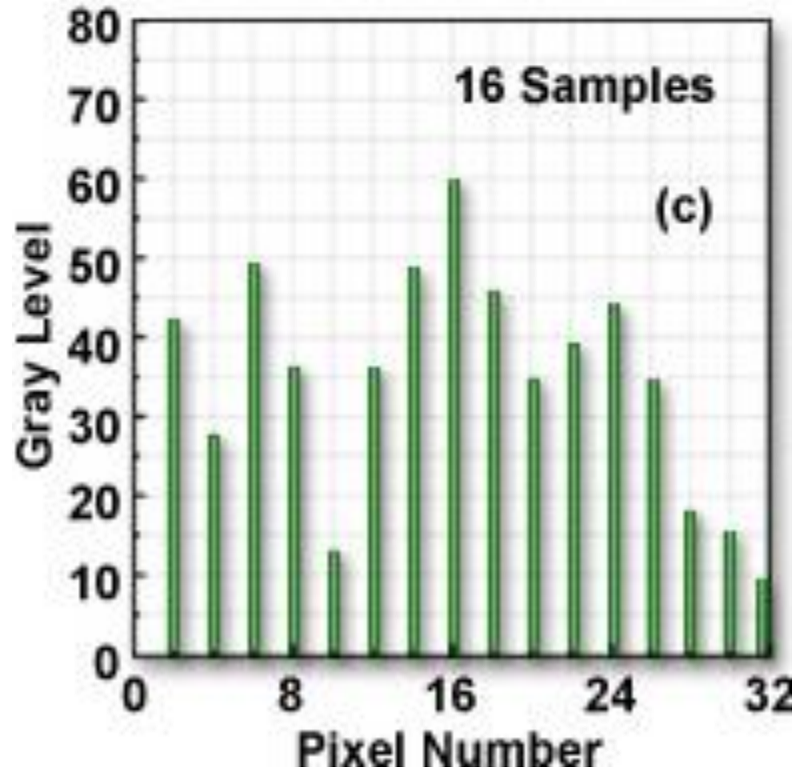




Or less

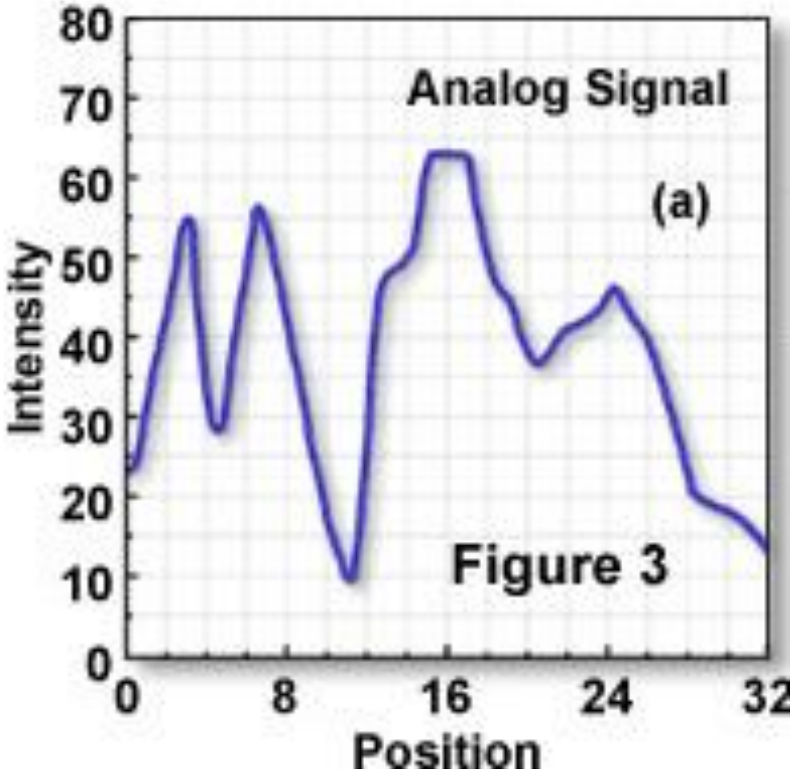


Analog Signal
(a)
Figure 3
Intensity
Position
0
10
20
30
40
50
60
70
80
0
8
16
24
32

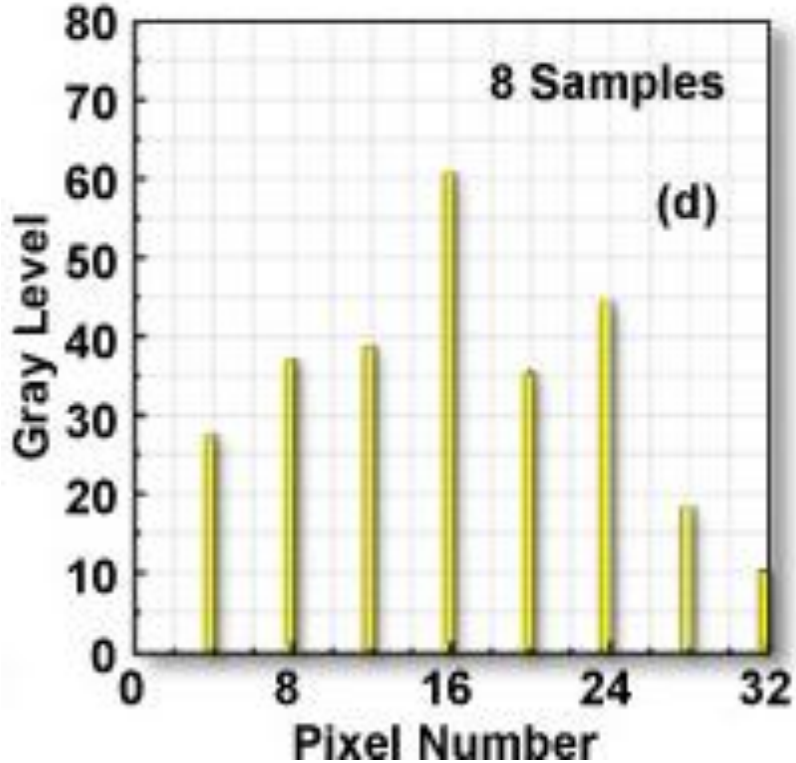


16 Samples
(c)
Gray Level
Pixel Number
0
10
20
30
40
50
60
70
80
0
8
16
24
32

Or even less



Analog Signal
(a)
Figure 3
Intensity
Position
80
70
60
50
40
30
20
10
0
0
8
16
24
32



8 Samples
(d)
Gray Level
Pixel Number
80
70
60
50
40
30
20
10
0
0
8
16
24
32

So, where's the tradeoff?

How often SHOULD we sample?

We need to examine two ideas to answer that question

Aliasing

Nyquist's Theorum

If a signal is undersampled,
it will be **aliased.**

Aliasing folds or wraps the signal, resulting in artifacts and distortions.

In the sound domain aliasing can appear as rough, dissonant, or spurious tones, or as noise.

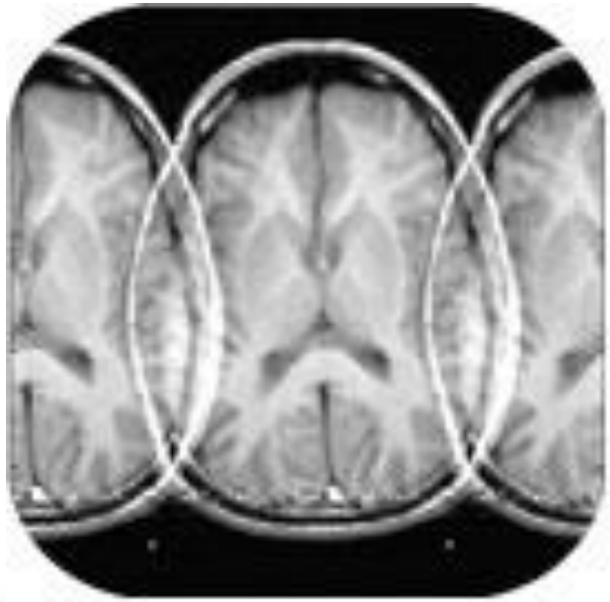
When spatial frequency is undersampled,

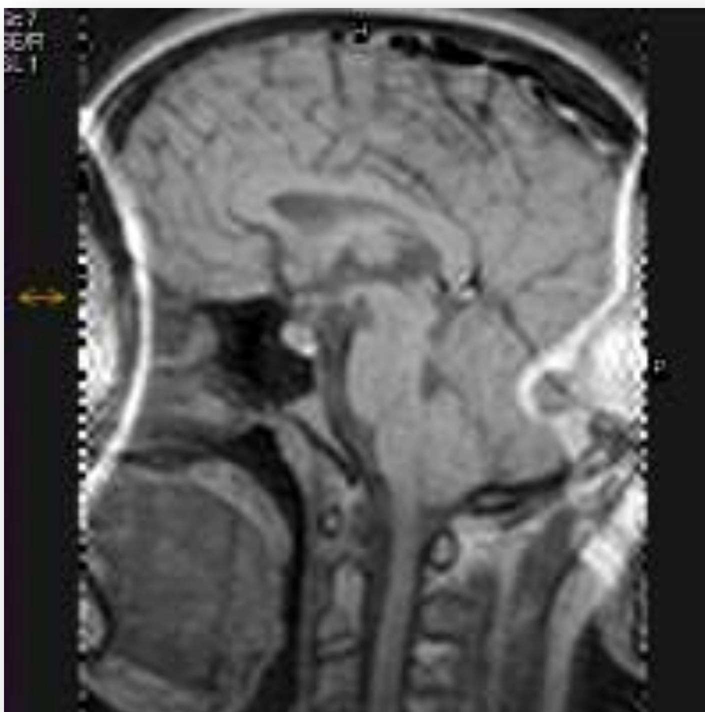
We see Moire patterns…

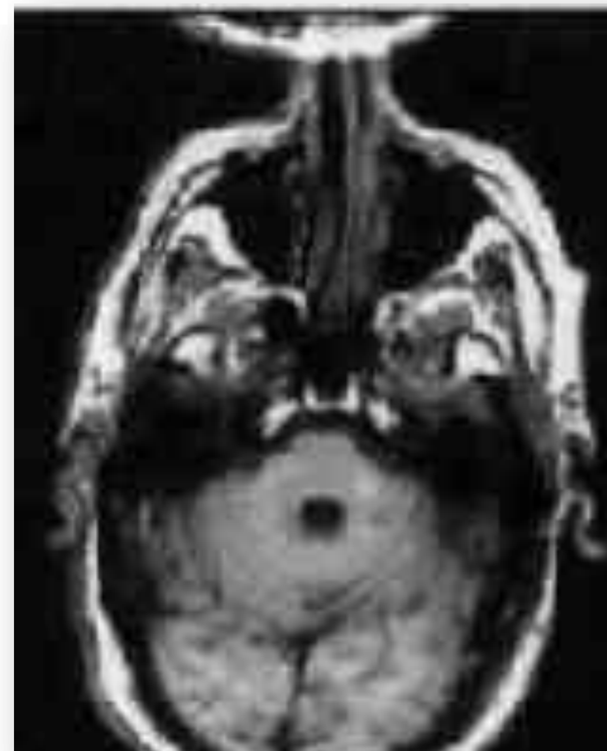
stair stepping etc.

In MR scans, aliasing occurs when any part of the imaged object extends outside the imaging volume.

We explicitly see the wrap around.







Nyquist's Theorum says that

Aliasing can be avoided if we sample at twice the bandwidth of the signal.

Usually this means twice the highest frequency in the signal.

For example, if the maximum frequency in a signal is 100 Hz,

We need to sample at 200 Hz (200 times per second) or more to prevent aliasing.

Aliasing in MRI images

Aliasing in MRI images can occur in the **frequency** or **phase** encoding direction.

To resolve aliasing in the "frequency encoding direction" we must have a high enough sample rate. So this is rarely an issue.

It is more difficult to correct aliasing in the "phase encoding direction" because phase space is 360 degrees*.

So, when you see aliasing in an MR image, it is usually in the phase encoding direction.

*For the incorrigibly curious: see notes for further explanation and reference, but more background is needed.

Nyquist's theorum is about sampling

frequency

…that is, sampling a cyclic event.

But not all aspects of a signal are cyclic.

In particular, amplitude is not cyclic,

So, Nyquist is not relevant to sampling amplitude.

Then, what do we do about digitizing amplitude?

Quantization is the process of digitizing amplitude information.

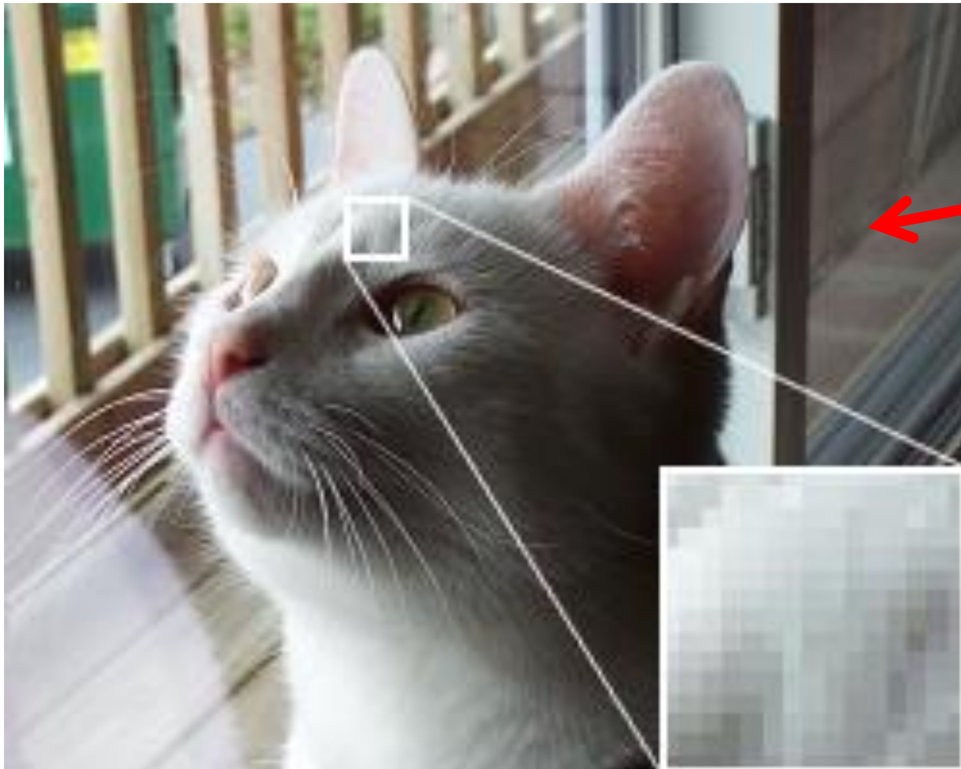
It involves approximating a continuous range of values with a **relatively-small set** of discrete values.

- similar to digitizing frequency, but with different rules

e.g., CD audio is quantized at 16 bits…

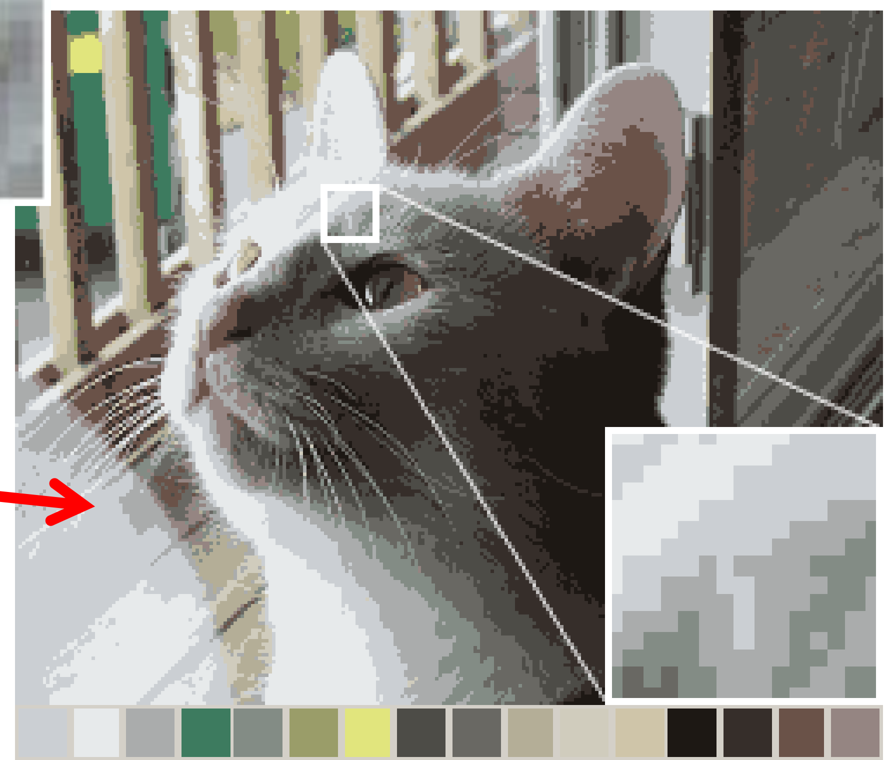
- …65,536 (i.e. 2^{16}) possible amplitude levels

In image processing, **quantization** may, for example, reduce the number of colors required to represent a digital image --



The 24 bit image… ~16 million+ colors (left)

-is quantized to 4 bits…16 colors (right)



making it possible to reduce file size, but at the expense of color range (bit depth).

Discrepencies in amplitude between the analog signal and the digital signal are called quantization noise.

Quantization noise occurs because of rounding errors that the computer inevitably makes in the process of digitizing a continuous signal.

Quantization noise is random.

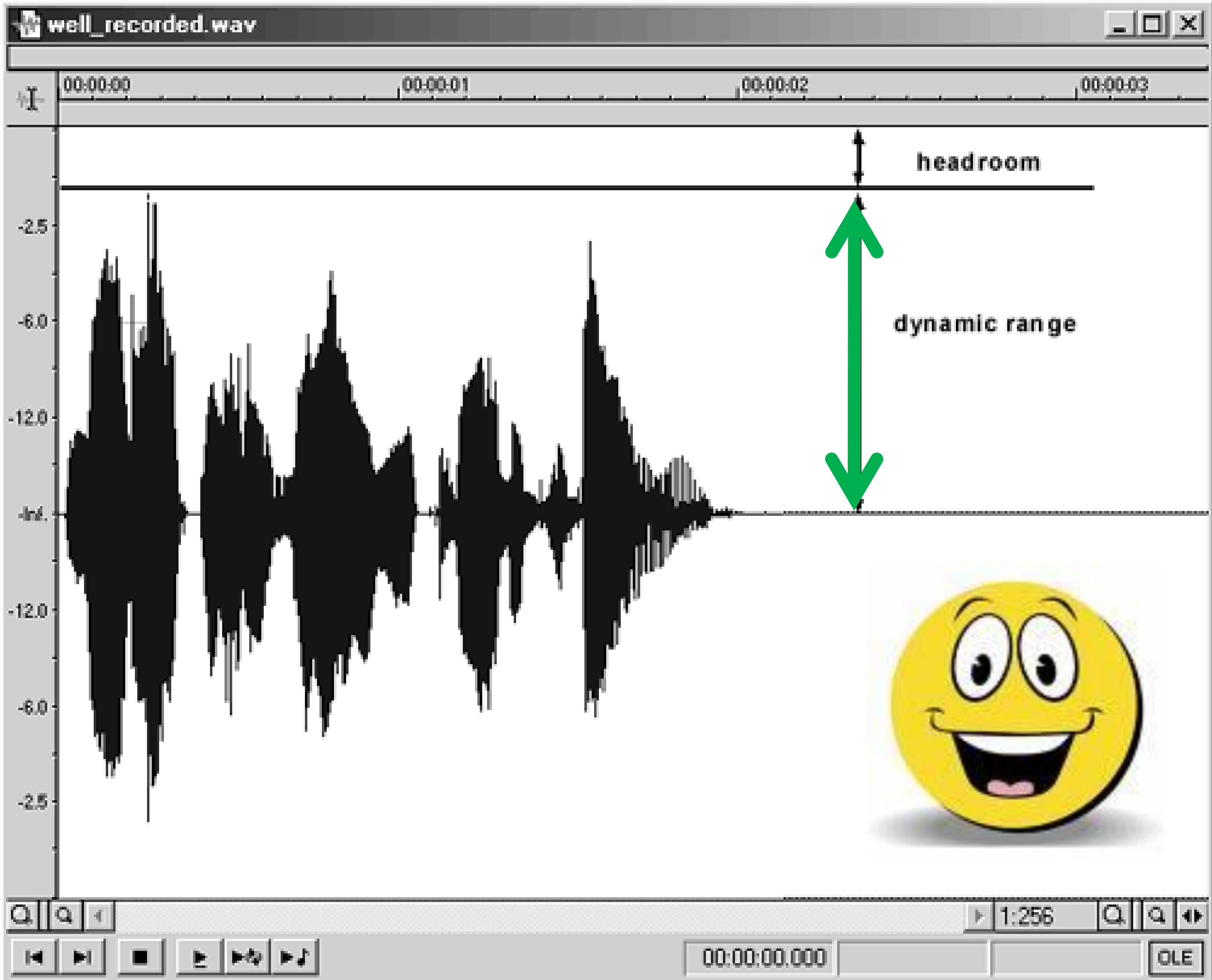
So, it is the same across the whole spectrum.

Thus the shape of the signal is preserved.

This means that quantization noise is **not as bad as aliasing**, because it doesn't distort the signal.

Dynamic range describes the ratio between the smallest and largest values of a signal.

Quantization is improved by matching the **dynamic range** of the digitized signal to the actual signal range.

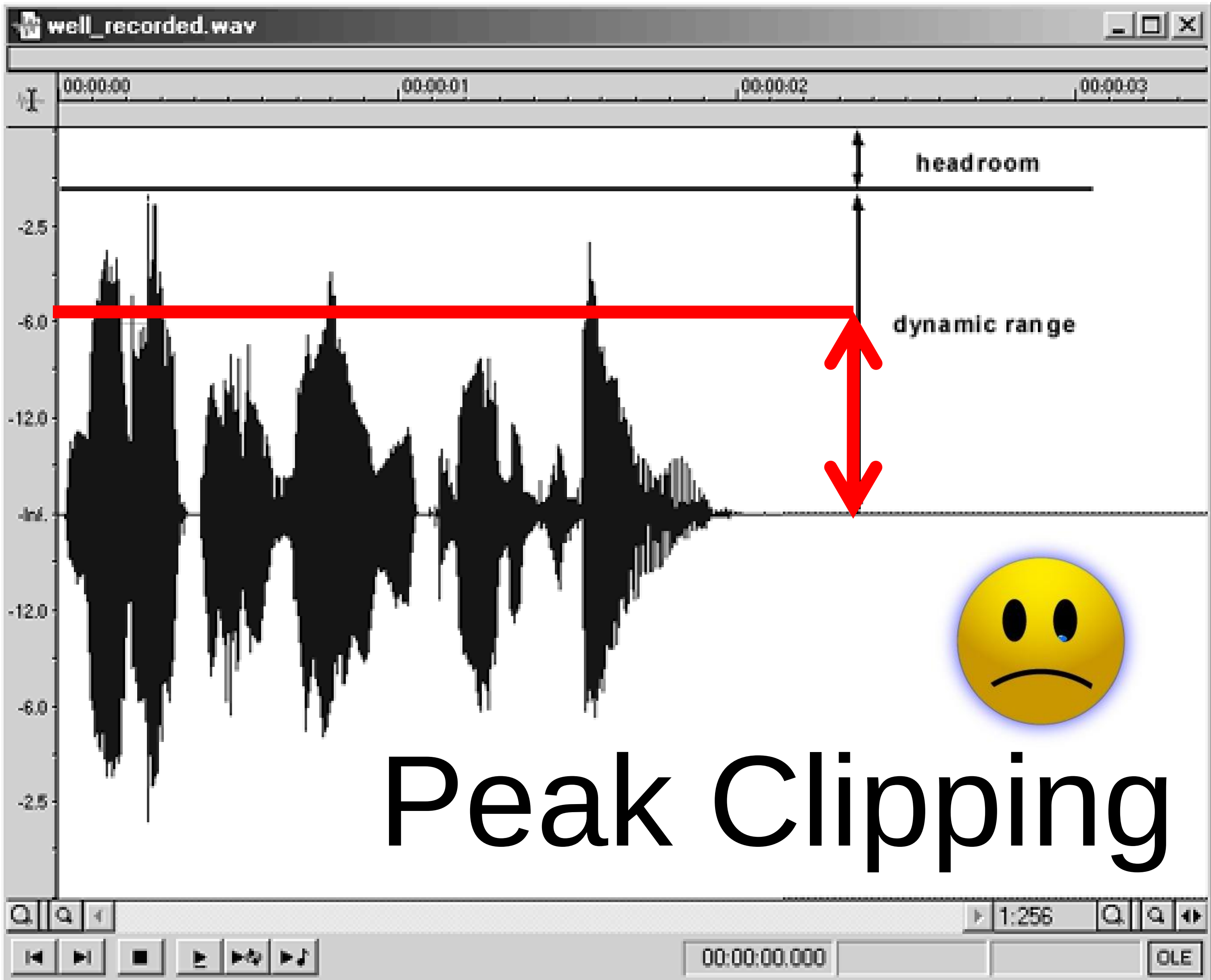

well_recorded.wav
00:00:00
00:00:01
00:00:02
00:00:03
headroom
dynamic range
-2.5
-6.0
-12.0
-Inf.
-12.0
-6.0
-2.5
1:256
00:00:00.000
OLE

Dynamic Range too Large

- If the dynamic range is larger than the signal range, then you create files that are larger than necessary, with no quality improvement.

Dynamic range too Small

- The range you sample must not be smaller than the actual signal range.
- When the dynamic sampling range is too small, it causes **peak clipping**


well_recorded.wav
00:00:00
00:00:01
00:00:02
00:00:03
headroom
dynamic range
-2.5
-6.0
-12.0
-Inf.
-12.0
-6.0
-2.5
1:256
00:00:00.000
OLE
Peak Clipping

Clipping

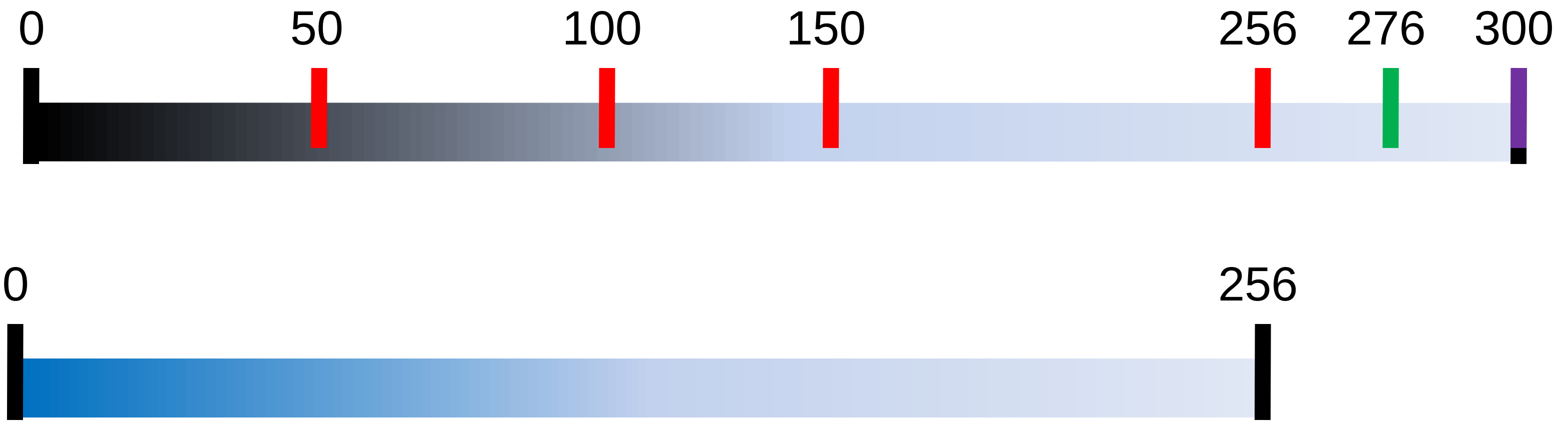
e.g., A system using 8-bit quantization, can represent 256 values.

If a signal contains values from 1 to 300

then all values over 256 in the original signal will be **clipped** at the maximum, 256.

Clipping

Original



Clipped Result

Image Clipping

Range: 0-255

Subtracted 100:
Range 0-155

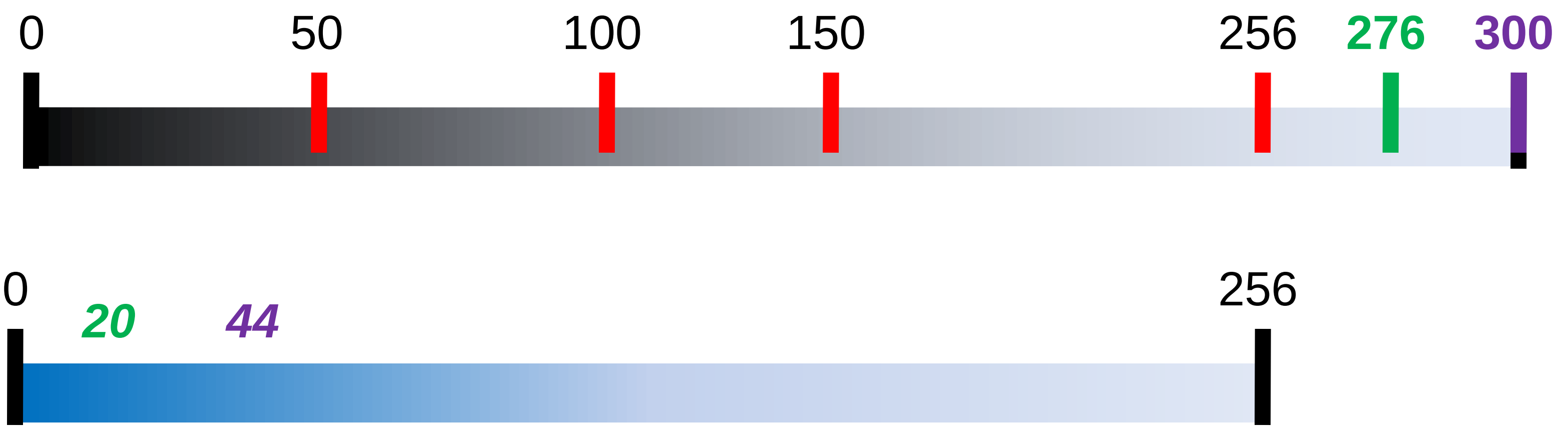
Added 100:
Range 100-255

Clipping is preferable to the alternative:

Wrapping (overflow), which results in gross signal distortion

If values overflowed/wrapped past 256, they would do this:

Original



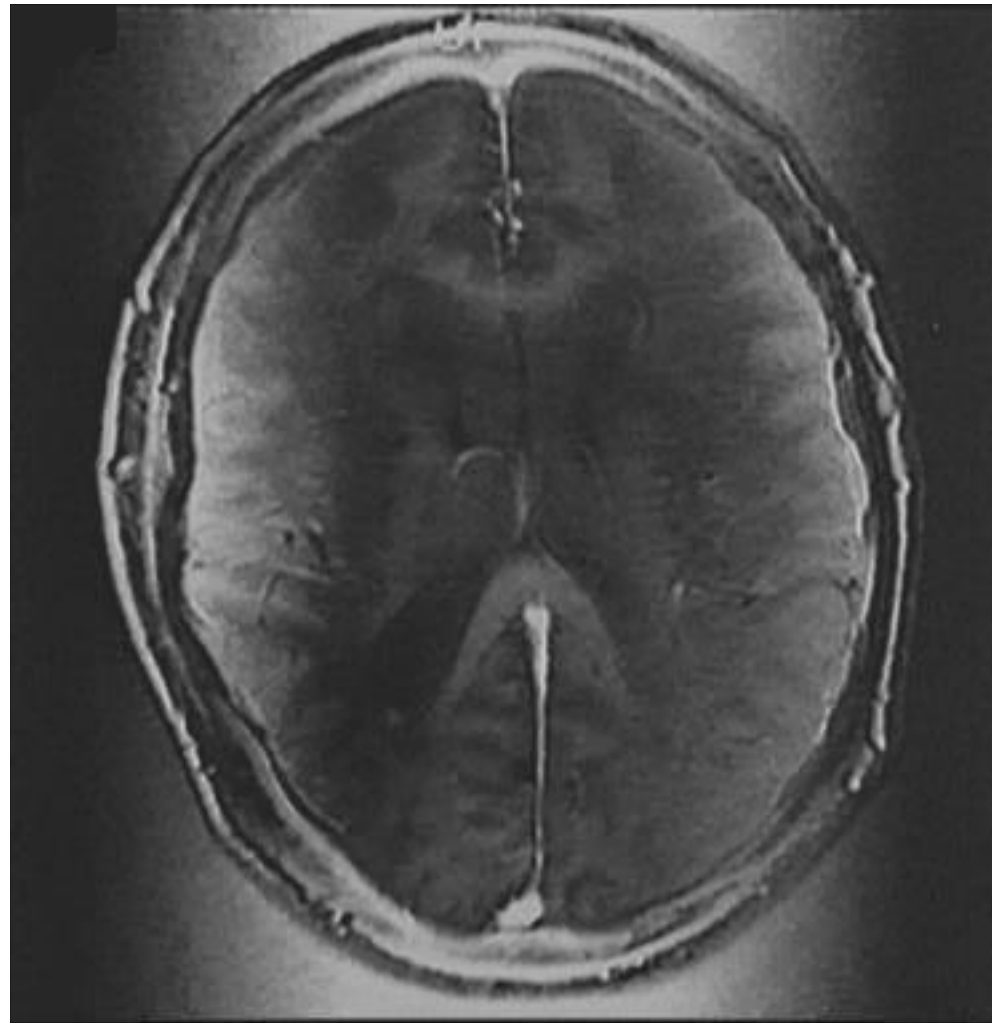
Wrapped Result

RF Overflow Artifact

Nonuniform, washed-out image.

Signal is too intense to be accurately digitized by Analog to Digital converter.

To fix it, one must adjust the receiver gain.



Let's Summarize

Digital Signal Processing is a fabulous extension of signal processing.

It depends on digitization.

Good digitization depends on getting sampling rate right.

Undersampling results in aliasing.

Correct sampling relies on Nyquist's theorum.

Quantization is digitization, measured in bits, which can be used to chop up
the non-time-dependent continuum, amplitude.

Amplitude quantization benefits from appropriate dynamic range.

A dynamic sampling range that is too small for the signal, results in peak clipping.

Quantization is never perfect, but quantization noise is not as serious as aliasing.

Signal Processing

Digital vs Analog

Digital **S**ignal **P**rocessing (DSP)

Sampling frequency (Rate)

Aliasing

Nyquist's Theorum

Quantization and Quantization Noise

Peak clipping and Dynamic Range

Overflow (Wrapping)