**Fundamentals of Digital Audio**

<table>
<thead>
<tr>
<th>Frequency</th>
<th>Order</th>
<th>Name 1</th>
<th>Name 2</th>
<th>Name 3</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 \cdot f = 440 Hz</td>
<td>n = 1</td>
<td>fundamental tone</td>
<td>1st harmonic</td>
<td>1st partial</td>
</tr>
<tr>
<td>2 \cdot f = 880 Hz</td>
<td>n = 2</td>
<td>1st overtone</td>
<td>2nd harmonic</td>
<td>2nd partial</td>
</tr>
<tr>
<td>3 \cdot f = 1320 Hz</td>
<td>n = 3</td>
<td>2nd overtone</td>
<td>3rd harmonic</td>
<td>3rd partial</td>
</tr>
<tr>
<td>4 \cdot f = 1760 Hz</td>
<td>n = 4</td>
<td>3rd overtone</td>
<td>4th harmonic</td>
<td>4th partial</td>
</tr>
</tbody>
</table>

![Diagram showing waveforms for different frequencies and orders](image)
Analog vs. Digital

**Analog**
- **Microphone**
- **Sound Pressure Wave**
- **Amplifier**
- **Voltage in Wire**
- **Sound Pressure Wave**

**Digital**
- **Analog to Digital Converter**
- **Computer Memory**
- **Digital to Analog Converter**
- **Amplifier**

Idealized Model
Sampling / Reconstruction

- **f(t)**
- **f*(t)**
- **quantizer**
- **computer**
- **g(nT)**
- **zero-order hold**
- **smoothing filter (low-pass)**
- **g(t)**

Model of the sampling and desampling process
Deconstruction of Sampling

Graphic representation of types of signals:

a) analog signal
b) quantized (continuous time) signal
c) sampled data signal
d) digital signal

Fidelity

depends on:

• sampling rate
• bit depth
• (encoding scheme)
**Sampling Rate**

**Human hearing:** 20-20,000 Hz

**Sampling Theorem:**
Must sample twice as high as highest component

**Typical rates:**
- 8 K dark / dull
- 11 K
- 22 K
- 44.1 K
- 48 K bright / clean

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**Fletcher & Munson**
Foldover in time domain

Positive and Negative Frequency

Nyquist frequency
**Digital Spectra**

quantization in time

- digital
- $|H(f)|$
- Nyquist Frequency $\frac{SR}{2}$
- sampling rate $SR$

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foldover in frequency domain

- foldover
- foldover frequency = $1/2$ sampling rate
- sampling rate

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Quantization in Amplitude

Distribution of Error

maximum error of ± 1/2
Computing S/N

“Ideal”

\[ dB = 20 \log \frac{2^N}{1} \]

or

\[ \frac{2^{N-1}}{0.5} \]

6 dB / bit

Bit Depth

Example S/N

<table>
<thead>
<tr>
<th>bits</th>
<th>dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>12</td>
</tr>
<tr>
<td>4</td>
<td>24</td>
</tr>
<tr>
<td>8</td>
<td>48 voice</td>
</tr>
<tr>
<td>12</td>
<td>72</td>
</tr>
<tr>
<td>16</td>
<td>96 CD</td>
</tr>
<tr>
<td>24</td>
<td>144 proaudio</td>
</tr>
</tbody>
</table>
Dynamic Range Problem

Low-level Signal Problem
Dither

Signal Reconstruction

Ideal reconstruction filter hasn't been found
**Oversampling Converters**

- **Multibit oversampling**
  convert 44.1 K to
  176.4 K (4x)
  352.8 K (8x)

- **1-bit oversampling**
  1 bit at high rate
  sigma-delta
  MASH
  \[128 \times 1\text{-bit} = 8 \times 16\text{-bits}\]
Actual Converters

• Combine oversampling first with 1-bit stream second

• Resulting noise lower
  4x \rightarrow 6\text{dB}
  8x \rightarrow 12\text{dB}

• In computers ground isolation of digital noise overwhelms everything else!