Reverberations:
Analog to Digital

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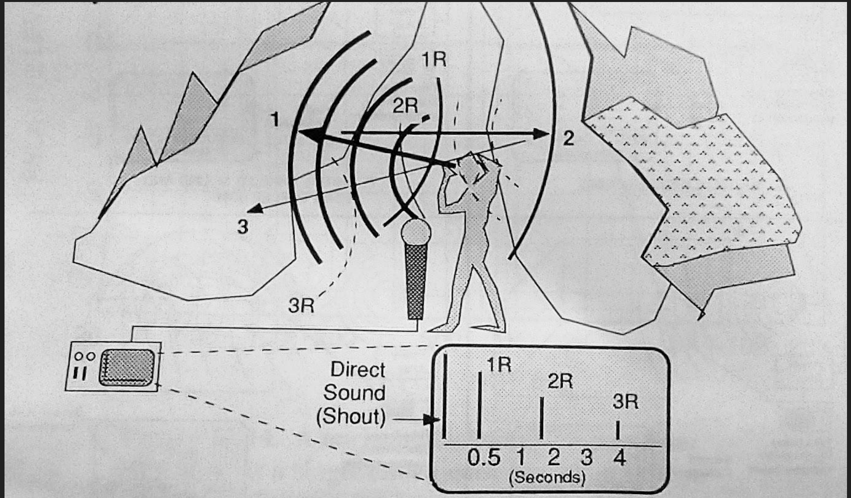
REVERBERANT SOUND BASICS

- Direct Sound vs. Echoes
- Reflected sound from environmental surfaces give the listener a sense of "space"



The first, second and third "contacts" of sound on the canyon walls are indicated by the numbered arrows. The subsequent reflections, as they reach the listner's ear (and the measurement mic), are numbered 1R, 2R and 3R. The third reflection is so weak as to barely be audible. These reflections are, in fact, echoes.

Figure 14-16. The natural occurence of echoes

Source: Yamaha Sound Reinforcement Handbook, 2nd Ed.

REVERB BASICS

- The length and density of the sound gives us cues of how big the environment is
- The uneven frequency response of the echoes give the space a timbral character
- Directionality of frequency components
  - Bass is omnidirectional, Treble is highly (uni-)directional
  - Our (stereo) ears use this directional information to perceive in 3 dimensions to pinpoint the source

REVERB BASICS

- Materials of reflected surfaces make a big difference

Source: Yamaha Sound Reinforcement Handbook, 2nd Ed.

MATERIAL	Frequency (Hz) 125	1k	4k
Brick Wall (18" Thick, unpainted)	.02	.04	.07
Brick Wall (18" Thick, painted)	.01	.02	.02
Interior Plaster (On metal lath)	.02	.06	.03
Poured Concrete	.01	.02	.03
Pine Flooring	.09	.08	.10
Carpeting (With pad)	.10	.30	.70
Drapes (Cotton, 2x fullness)	.07	.80	.50
Drapes (Velour, 2x fullness)	.15	.75	.65
Acoustic Tile (5/8", #1 Mount*)	.15	.70	.65
Acoustic Tile (5/8", #2 Mount*)	.25	.70	.65
Acoustic Tile (5/8", #7 Mount*)	.50	.75	.65
Tectum Panels (1", #2 Mount*)	.08	.55	.65
Tectum Panels (1", #7 Mount*)	.35	.35	.65
Plywood Panel (1/8", 2" Air space)	.30	.10	.07
Plywood Cylinders (2 Layers, 1/8")	.35	.20	.18
Perforated Transite (w/Pad, #7 Mount*)	.90	.95	.45
Occupied Audience Seating Area	.50	.95	.85
Upholstered Theatre Seats (Hard Floor)	.45	.90	.70

* #1 Mount is cemented directly to plaster or concrete,
#2 Mount is fastened to nominal 1" thick furring strips,
#7 Mount is suspended ceiling w/ 16" air space above.

Table 6-1. Approximate absorption coefficients of common materials

REVERB BASICS

RT60 is the standard measurement of reverb time

- the time required for the reverb signal to decay away to 1/1000th (-60 dB) of the original signal strength

	MKS units: S = Surface area in m^2 V= Volume in m^3	English units: S = Surface area in ft^2 V= Volume in ft^3
Sabine Gives best correspondence with published absorption coefficients where $\bar{\alpha}<0.2$.	$T = \frac{0.16V}{S\bar{\alpha}}$	$T = \frac{0.49V}{S\bar{\alpha}}$
Eyring Preferred formula for well-behaved rooms having $\bar{\alpha} \gtrsim 0.2$.	$T = \frac{0.16V}{-S \ln(1-\bar{\alpha})}$	$T = \frac{0.49V}{-S \ln(1-\bar{\alpha})}$
Fitzroy For rectangular rooms with poorly distributed absorption. α_x, α_y and α_z are average absorption coefficients of opposing pairs of surfaces with total areas of x, y and z.	$T = \frac{0.16V}{S^2}\left(\frac{x^2}{X\alpha_x} + \frac{y^2}{Y\alpha_y} + \frac{z^2}{Z\alpha_z}\right)$	$T = \frac{0.49V}{S^2}\left(\frac{x^2}{X\alpha_x} + \frac{y^2}{Y\alpha_y} + \frac{z^2}{Z\alpha_z}\right)$
	T = Decay time (in seconds) for 60 dB level reduction.	

Figure 6-6. Reverberation time equations

Source: Yamaha Sound Reinforcement Handbook, 2nd Ed.

REVERB BASICS

- In this talk, I will go into some detail about how we have faked this effect, with both analog and digital devices

CHAMBER REVERB

1930's

- Used in radio production / recording studios
  - A speaker and a microphone in a "chamber" or room.
  - Aimed facing away from each other to minimize direct sound
- Completely ordinary spaces worked as well
  - Tiled bathrooms
  - Hallways
  - Stairwells

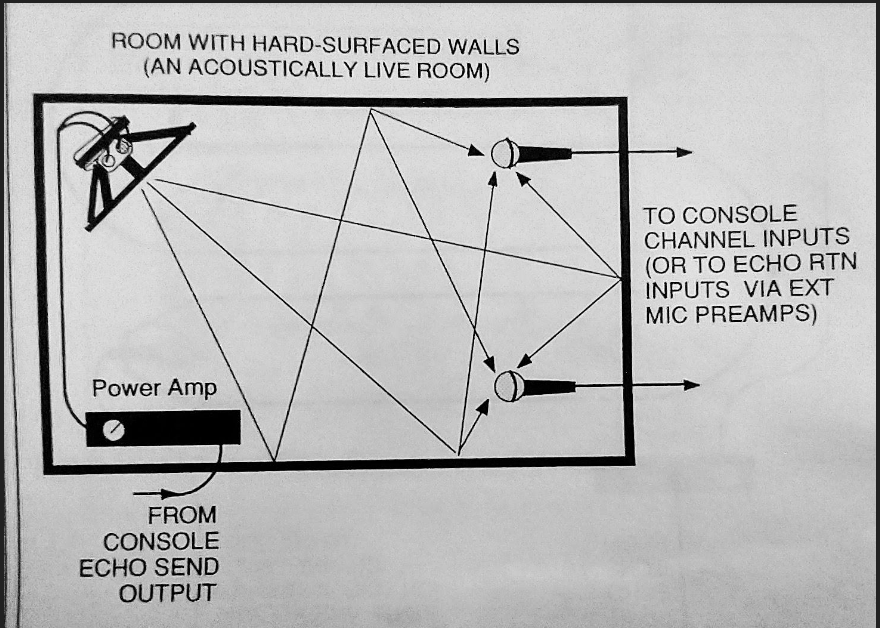


Figure 14-17. General design of an actual reverb chamber

Source: Yamaha Sound Reinforcement Handbook, 2nd Ed.

CHAMBER REVERB

- Non-parallel walls - avoid frequency problems and standing waves
- Hard surfaces work best for long decay times
- An investment in construction & real estate

PLATE REVERB

1957 - EMT 140

- Better high-frequency detail
- Smaller than a chamber
- 600+ lbs
- dimensions roughly 4' x 8' x 1'

PLATE REVERB

- Dispersion: sound moves faster for higher frequencies
- Damping plate, controlled by a servo motor, allowed adjustment of the reverb time (up to 4 sec)
- Stereo imaging: contact pickups can be in stereo, and they are not equidistant from the driver
- Sound waves reflect when they reach the edges

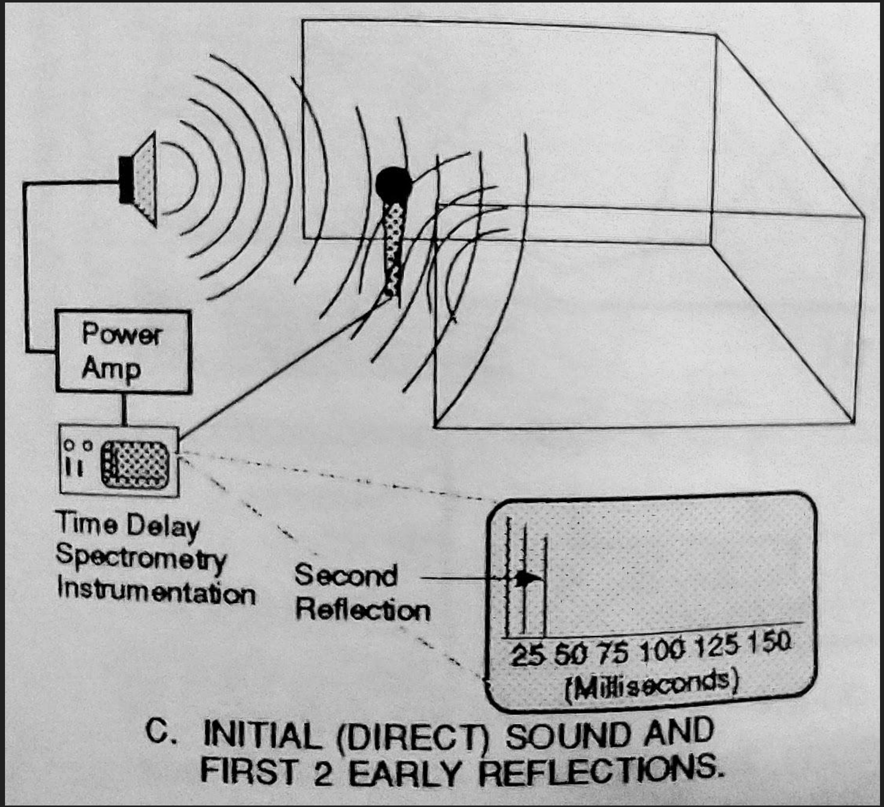
PLATE REVERB

PLATE REVERB

Criticisms:

- Has a very dense sound and needs a predelay (or added *early reflections*).
- Tape delay or the Binson Echo machine was used for predelay to make it sound more "realistic".

Source: Yamaha Sound Reinforcement Handbook, 2nd Ed.



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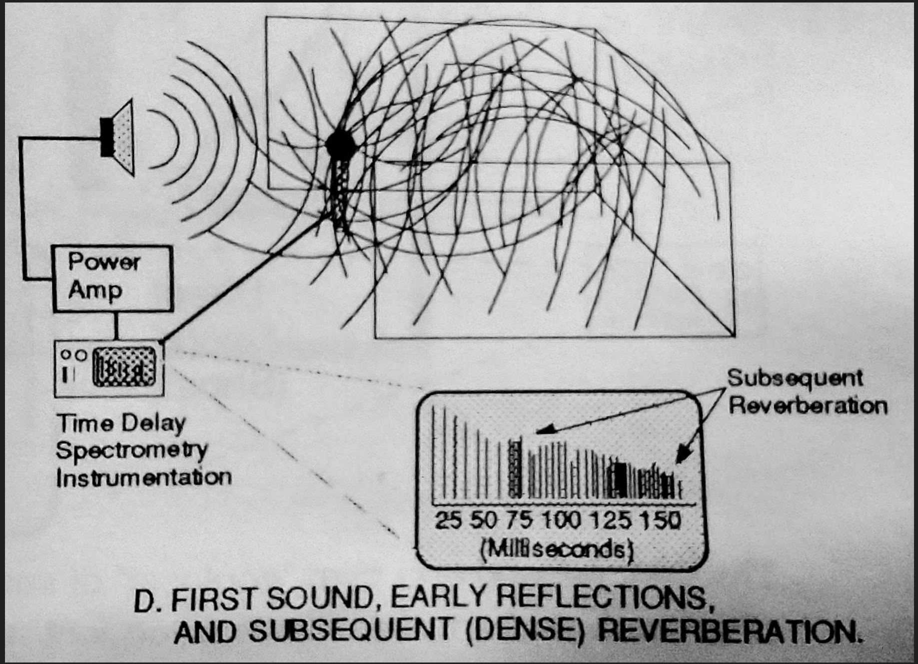
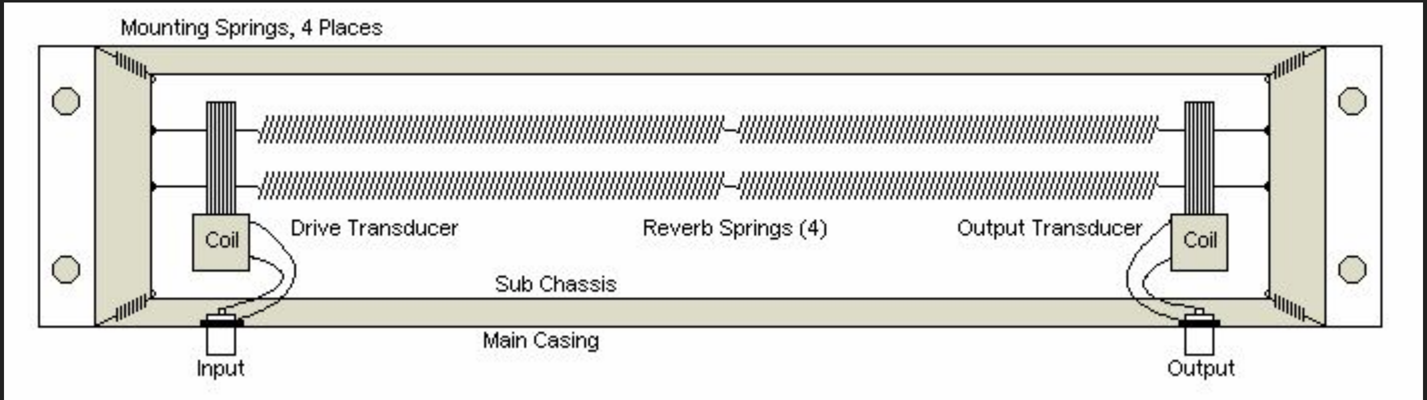


Source: Yamaha Sound Reinforcement Handbook, 2nd Ed.

SPRING REVERB

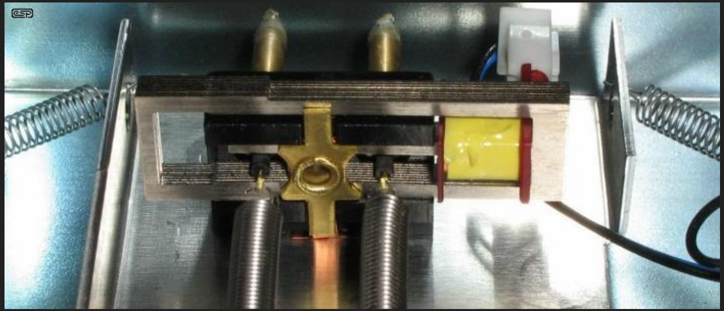
1960 - Hammond Organ - 1st commercially available Spring Reverb

- Why ORGAN? Many churches acoustically deadened to allow intelligible speech - put that grandeur back into the sound
- Accutronics licensed to Leo Fender 1963, used in the 1963 Fender Vibroverb amp



SPRING REVERB

- Literally, a pair (or more) of metal springs connected to output and input transducers. Radial vibrations picked up by the other end.
- You can mix the effected signal into the dry, uneffected signal via knob
- Light, portable, doesn't add much weight to a guitar amplifier. Aluminum housing and steel springs, simple construction, & **CHEAP**.



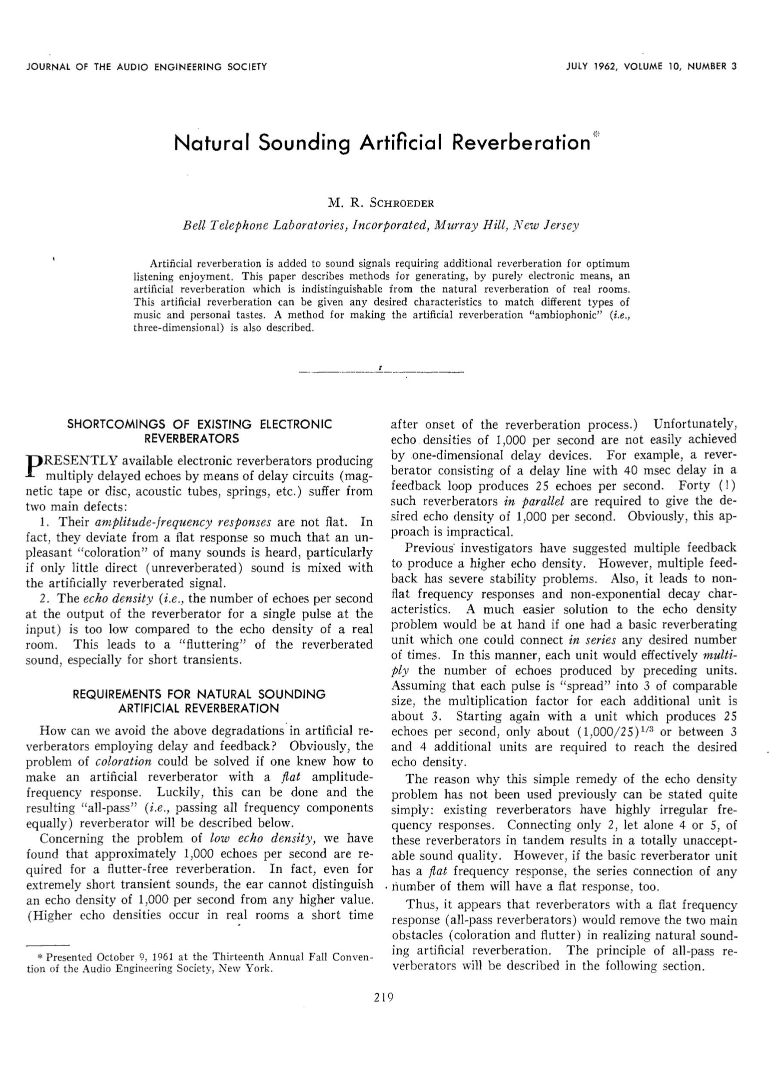
DIGITAL REVERB

Manfred Schroeder

1962 AES Paper, “Natural Sounding Artificial Reverberation”

Artificial reverberation is added to sound signals requiring additional reverberation for optimum listening enjoyment. This paper describes methods for generating, by purely electronic means, an artificial reverberation which is indistinguishable from the natural reverberation of real rooms. This artificial reverberation can be given any desired characteristics to match different types of music and personal tastes. A method for making the artificial reverberation - ambiophonic- (i.e., three-dimensional) is also described.

Author: Schroeder, Manfred R.
Affiliation: Bell Telephone Laboratories, Incorporated, Murray Hill, NJ
JAES Volume 10 Issue 3 pp. 219-223; July 1962
Publication Date: July 1, 1962 Import into BibTeX
Permalink: http://www.aes.org/e-lib/browse.cfm?elib=849



JOURNAL OF THE AUDIO ENGINEERING SOCIETY JULY 1962, VOLUME 10, NUMBER 3

Natural Sounding Artificial Reverberation*

M. R. SCHROEDER
Bell Telephone Laboratories, Incorporated, Murray Hill, New Jersey

Artificial reverberation is added to sound signals requiring additional reverberation for optimum listening enjoyment. This paper describes methods for generating, by purely electronic means, an artificial reverberation which is indistinguishable from the natural reverberation of real rooms. This artificial reverberation can be given any desired characteristics to match different types of music and personal tastes. A method for making the artificial reverberation “ambiophonic” (*i.e.*, three-dimensional) is also described.

SHORTCOMINGS OF EXISTING ELECTRONIC REVERBERATORS

PRESENTLY available electronic reverberators producing multiply delayed echoes by means of delay circuits (magnetic tape or disc, acoustic tubes, springs, etc.) suffer from two main defects:

1. Their *amplitude-frequency responses* are not flat. In fact, they deviate from a flat response so much that an unpleasant “coloration” of many sounds is heard, particularly if only little direct (unreverberated) sound is mixed with the artificially reverberated signal.
2. The *echo density* (*i.e.*, the number of echoes per second at the output of the reverberator for a single pulse at the input) is too low compared to the echo density of a real room. This leads to a “fluttering” of the reverberated sound, especially for short transients.

REQUIREMENTS FOR NATURAL SOUNDING ARTIFICIAL REVERBERATION

How can we avoid the above degradations in artificial reverberators employing delay and feedback? Obviously, the problem of *coloration* could be solved if one knew how to make an artificial reverberator with a *flat* amplitude-frequency response. Luckily, this can be done and the resulting “all-pass” (*i.e.*, passing all frequency components equally) reverberator will be described below.

Concerning the problem of *low echo density*, we have found that approximately 1,000 echoes per second are required for a flutter-free reverberation. In fact, even for extremely short transient sounds, the ear cannot distinguish an echo density of 1,000 per second from any higher value. (Higher echo densities occur in real rooms a short time after onset of the reverberation process.) Unfortunately, echo densities of 1,000 per second are not easily achieved by one-dimensional delay devices. For example, a reverberator consisting of a delay line with 40 msec delay in a feedback loop produces 25 echoes per second. Forty (!) such reverberators *in parallel* are required to give the desired echo density of 1,000 per second. Obviously, this approach is impractical.

Previous investigators have suggested multiple feedback to produce a higher echo density. However, multiple feedback has severe stability problems. Also, it leads to non-flat frequency responses and non-exponential decay characteristics. A much easier solution to the echo density problem would be at hand if one had a basic reverberating unit which one could connect *in series* any desired number of times. In this manner, each unit would effectively *multiply* the number of echoes produced by preceding units. Assuming that each pulse is “spread” into 3 of comparable size, the multiplication factor for each additional unit is about 3. Starting again with a unit which produces 25 echoes per second, only about $(1{,}000/25)^{1/3}$ or between 3 and 4 additional units are required to reach the desired echo density.

The reason why this simple remedy of the echo density problem has not been used previously can be stated quite simply: existing reverberators have highly irregular frequency responses. Connecting only 2, let alone 4 or 5, of these reverberators in tandem results in a totally unacceptable sound quality. However, if the basic reverberator unit has a *flat* frequency response, the series connection of any number of them will have a flat response, too.

Thus, it appears that reverberators with a flat frequency response (all-pass reverberators) would remove the two main obstacles (coloration and flutter) in realizing natural sounding artificial reverberation. The principle of all-pass reverberators will be described in the following section.

* Presented October 9, 1961 at the Thirteenth Annual Fall Convention of the Audio Engineering Society, New York.

DIGITAL REVERB

Schroeder describes 2 algorithms:

1. 4 or more parallel comb filters (delay lines with feedback) of unequal length, with the outputs summed and run through two or more allpass delays.
2. Describes a reverberator using 5 allpass delays in series.

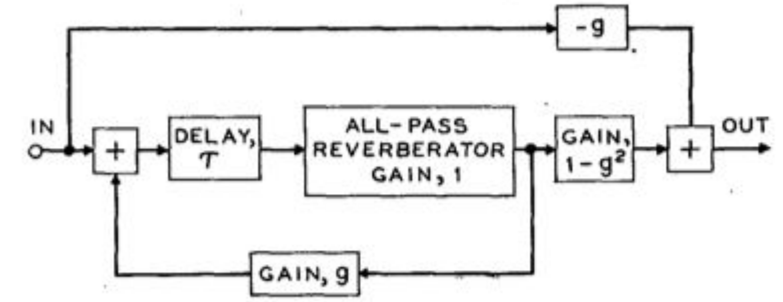


FIG. 5. All-pass reverberator with variable ratio of direct-to-reverberated sound. It produces a non-exponential decay of the reverberated sound.

DIGITAL REVERB

Allpass Filters are a big innovation due to **flat frequency response**

By placing the allpasses in a feedback path, a much more natural reverberation decay is created. Schroeder's paper is the first description of a nested allpass delay. The idea of putting allpasses inside of delayed feedback loops is fundamental to the algorithms of Lexicon, Alesis, and other high end reverberator manufacturers, and is still used to this day.

- These algorithms were run on MAINFRAMES and were so slow and computationally expensive they took DAYS to run

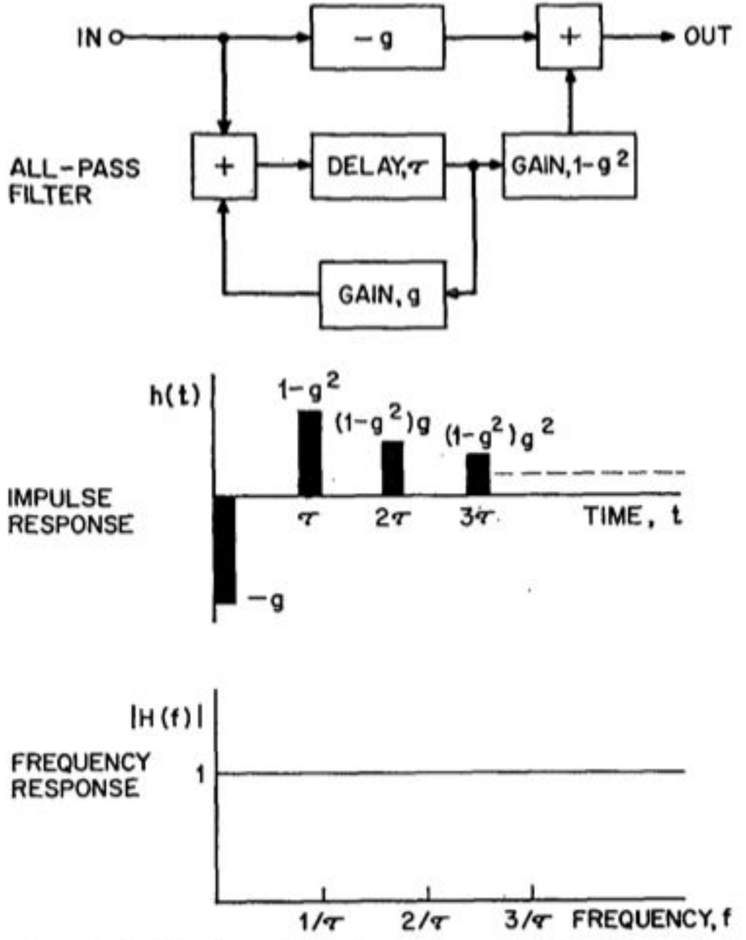


FIG. 2. Modification of simple reverberator. By adding proper amount of undelayed signal, frequency response of the reverberator becomes flat (all-pass reverberator).

DIGITAL IN COMMERCIAL HARDWARE

1976 - Dr. Barry Blesser

EMT 250 - first digital reverb hardware unit

- 16K memory chips
- $15,000 MSRP in 1976 is equivalent in purchasing power to about $68,958.35 today
- Up to 10 seconds of decay

DIGITAL IN COMMERCIAL HARDWARE

1978 - Lexicon 224

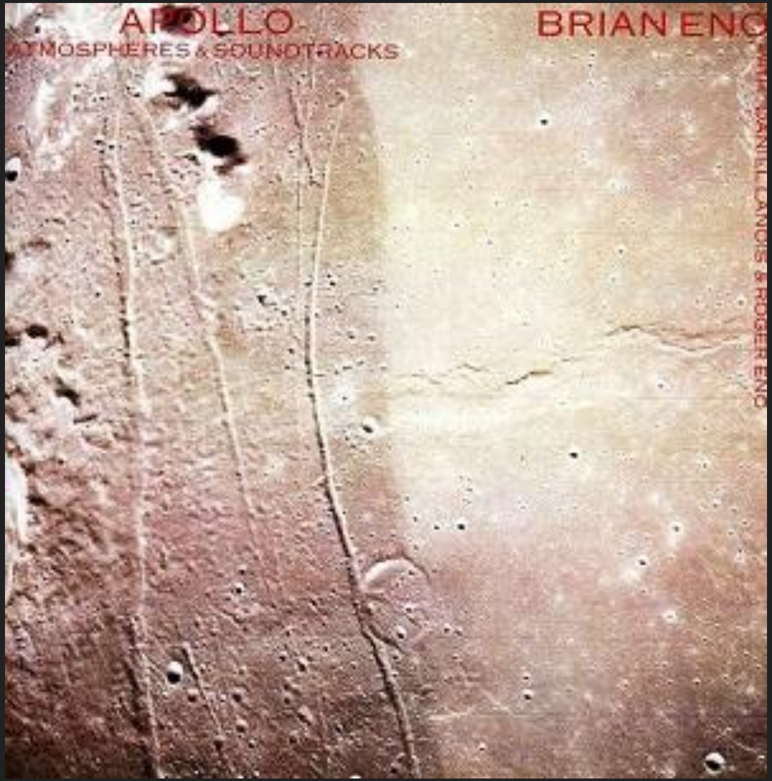
Dr. David Griesinger, a nuclear physicist, musician, classical recording engineer

- half the price of the EMT 250
- up to 70 seconds of decay

DIGITAL IN COMMERCIAL HARDWARE

Increased reverb time equals

AMBIENT TIME



Cascading Comb Filter Banks

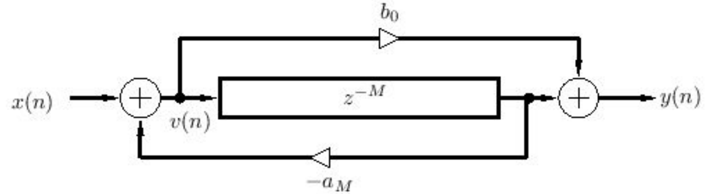
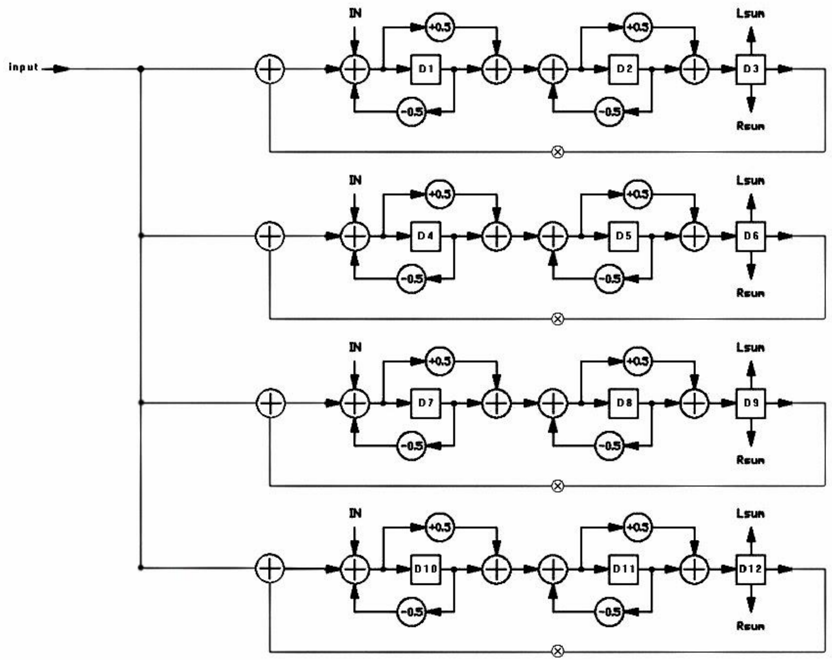


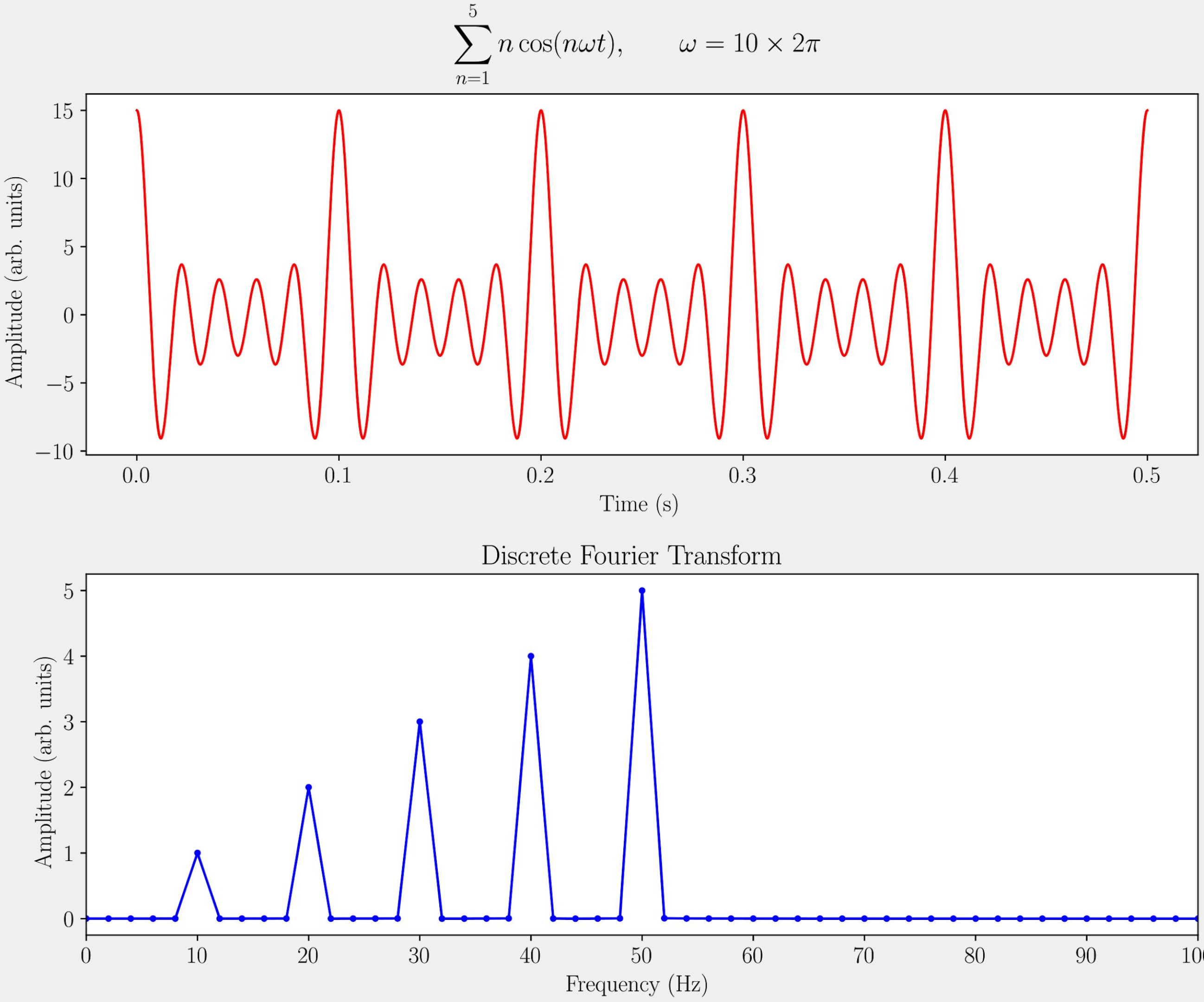
Figure 2.30: A combined feedback/feedforward comb filter which gives an allpass filter when $b_0 = a_M$.



Still Used Today!!!

<u>**Section 2:**</u>

CONVOLUTION
REVERB
and the
DISCRETE
FOURIER
TRANSFORM

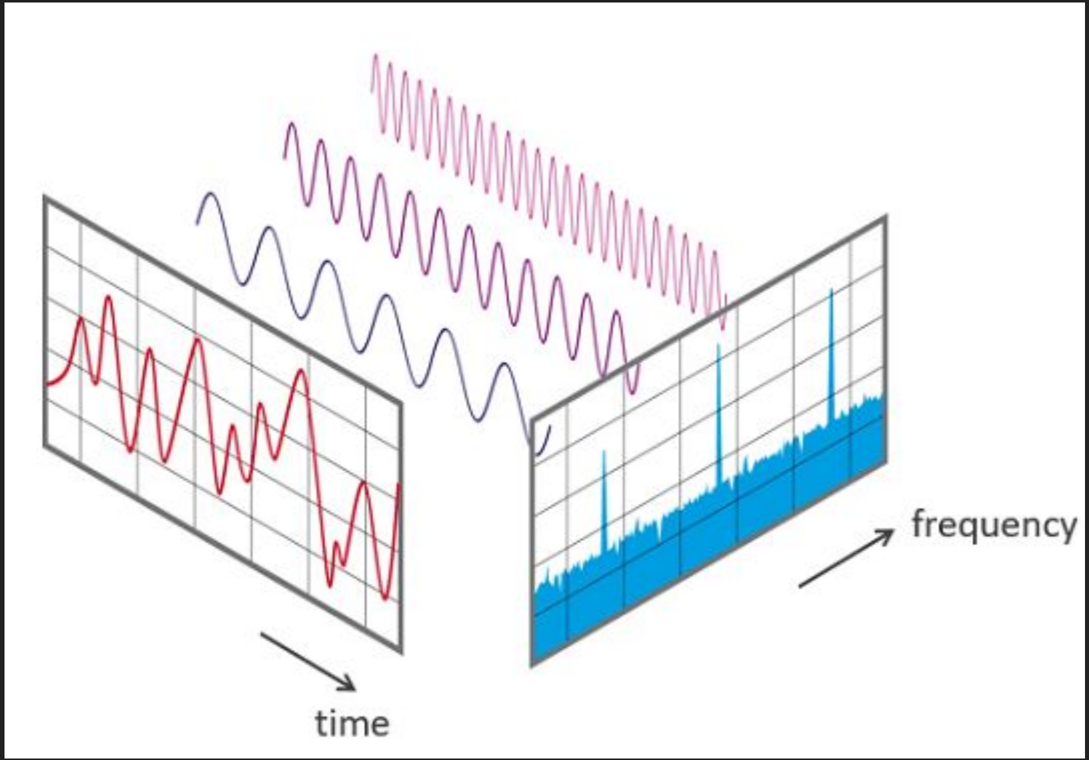


Fourier Analysis

1807

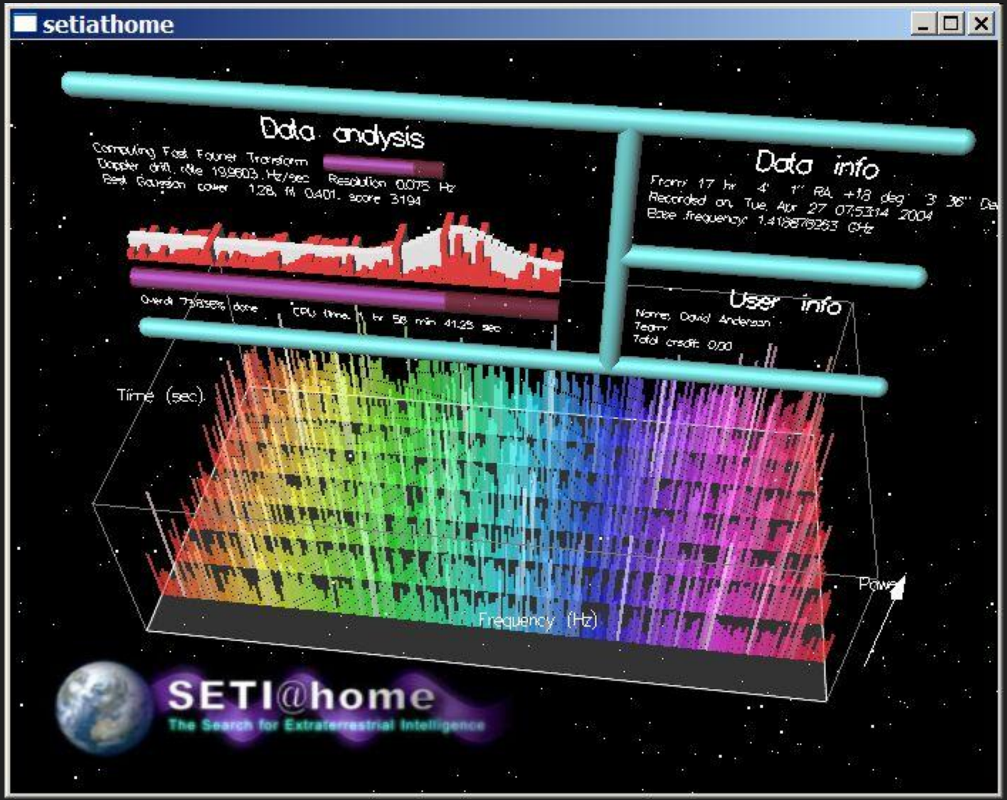
Jean-Baptiste Fourier

best known for initiating the investigation of Fourier series, which eventually developed into Fourier analysis and harmonic analysis, and their applications to problems of heat transfer and vibrations.



SETI@home, UC Berkeley

- Using home PCs to find aliens
- Look at all these cool radio wiggles Arecibo recorded
- Regular pulses might mean broadcasts originating from intelligent life



Towards CONVOLUTION REVERB

1962 Karatsuba and Ofman published an algorithm for fast multiplication of large numbers with many digits, widely known as the Karatsuba algorithm. For several decades, the algorithm was the fastest known multiplication algorithm for practical problem sizes.

- Important tool for the design of hardware multipliers in integrated circuits.
- Although the original Karatsuba algorithm was originally conceived for fast multiplication of integers, it can also be applied to other algebraic structures, allowing fast polynomial multiplication and as well fast convolution.

The Karatsuba algorithm is a classic example for a divide-and-conquer strategy.

- Let $x(n)$ and $h(n)$ be two length-2N sequences.
- Both sequences are split in half, forming four length-N subsequences
  $x0(n) = x(0),...,x(N-1)$ $\qquad$ $x1(n) = x(N),...,x(2N-1)$
  $h0(n) = h(0),...,h(N-1)$ $\qquad$ $h1(n) = h(N),...,h(2N-1)$

Towards CONVOLUTION REVERB

Karatsuba and Ofman

The $2N \times 2N$-linear convolution $x(n) * h(n)$ can be expressed using four $N \times N$-sub convolutions:

$$y0(n) = x0(n) * h0(n)$$
$$y1(n) = x0(n) * h1(n) + x1(n) * h0(n)$$
$$y2(n) = x1(n) * h1(n)$$

OVERLAP-ADD

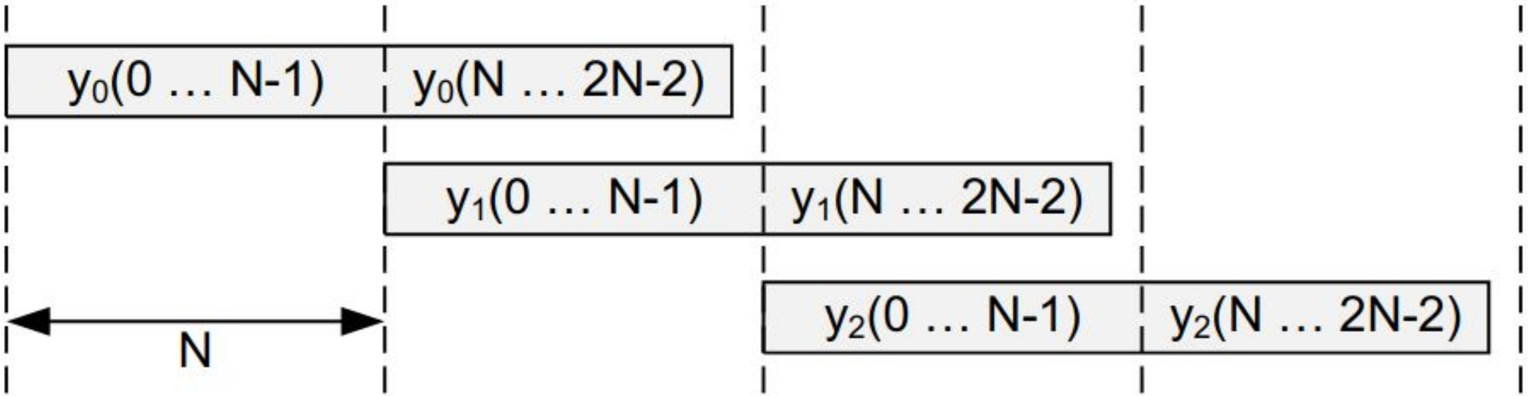


Figure 2.4.: Overlap-Add scheme used in the Karatsuba convolution algorithm

FUN FACTS about Fast Fourier Transform

Cooley-Tukey - 1965 “twiddle-factor FFT paper”

Carl Friedrich Gauss (1777-1855) can be considered the first inventor of the FFT. Gauss developed an algorithm which is mathematically equivalent to the *Cooley-Tukey twiddle-factor FFT*; it was never published in his lifetime and only found posthumously - he used it for computing the trajectories of asteroids.

Gauss quotes I might have accidentally said:

- Mathematicians stand on each other's shoulders.
- It is not knowledge, but the act of learning, not possession but the act of getting there, which grants the greatest enjoyment.
- The enchanting charms of this sublime science reveal only to those who have the courage to go deeply into it.
- I have had my results for a long time: but I do not yet know how I am to arrive at them.

DISCRETE FOURIER TRANSFORM

continuous to discrete time-series so we can use DIGITAL SAMPLES of audio

$$\boxed{f(k) = \int_{-\infty}^{\infty} f(x)e^{-2\pi ikx}dx} \xrightarrow{\textbf{Discretize}} \boxed{f_k = \Sigma_0^{N-1} x_n e^{\frac{-2\pi ikn}{N}}}$$

FT **DFT**

FFT-based fast convolution

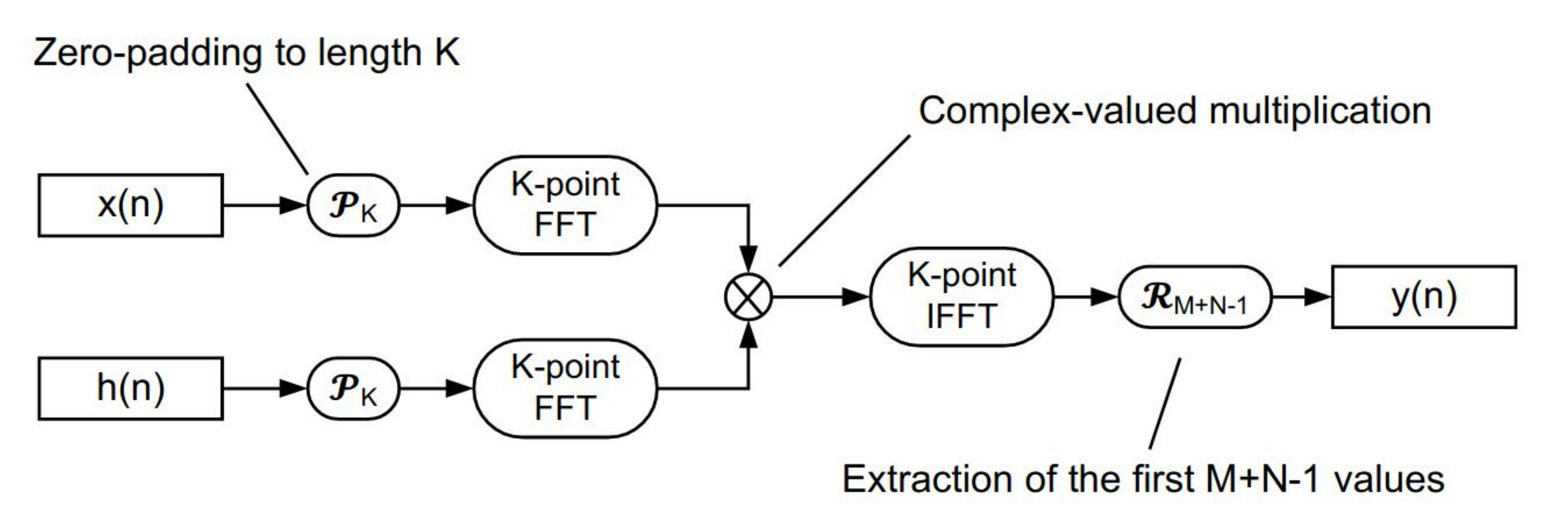
- Let $x(n)$ be a **length-M** and $h(n)$ a **length-N** sequence.
- The $M \times N$ linear convolution $x(n) * h(n)$ can be computed in $O(K \log K)$, using K-point FFTs. The transform size K must satisfy $K \geq M + N - 1$, otherwise the linear convolution is corrupted. **K should be a power of 2**, and **the sequence should be zero-padded to the length of the next power of 2 if not**.

Karatsuba + DFT

- The next length-B input signal block is zero-padded to length-K and transformed using a K-point FFT.
- The length-N impulse response is zero-padded to length-K and transformed using a K-point FFT.
- The input and filter DFT coefficients are pair-wisely multiplied (**spectral convolution**)
- The result is transformed back into the time-domain using a K-point IFFT. It forms a partial convolution result of length B+N−1, which is buffered.
- The length-B output block is **added up from the overlapping partial results** (Overlap-Add step).

Putting it all together...

We can use OVERLAP-ADD to form our output signal.



Overlap-Add partitions the INPUTS

3. *Partitioned convolution techniques*

$$
\begin{bmatrix} y_0 \\ y_1 \\ y_2 \\ y_3 \\ y_4 \\ y_5 \\ y_6 \\ y_7 \\ y_8 \\ y_9 \\ y_{10} \\ y_{11} \end{bmatrix}
=
\begin{bmatrix}
h_0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\
h_1 & h_0 & 0 & 0 & 0 & 0 & 0 & 0 \\
h_2 & h_1 & h_0 & 0 & 0 & 0 & 0 & 0 \\
h_3 & h_2 & h_1 & h_0 & 0 & 0 & 0 & 0 \\
h_4 & h_3 & h_2 & h_1 & h_0 & 0 & 0 & 0 \\
0 & h_4 & h_3 & h_2 & h_1 & h_0 & 0 & 0 \\
0 & 0 & h_4 & h_3 & h_2 & h_1 & h_0 & 0 \\
0 & 0 & 0 & h_4 & h_3 & h_2 & h_1 & h_0 \\
0 & 0 & 0 & 0 & h_4 & h_3 & h_2 & h_1 \\
0 & 0 & 0 & 0 & 0 & h_4 & h_3 & h_2 \\
0 & 0 & 0 & 0 & 0 & 0 & h_4 & h_3 \\
0 & 0 & 0 & 0 & 0 & 0 & 0 & h_4
\end{bmatrix}
\cdot
\begin{bmatrix} x_0 \\ x_1 \\ x_2 \\ x_3 \\ x_4 \\ x_5 \\ x_6 \\ x_7 \end{bmatrix}
$$

Figure 3.1.: Overlap-Add convolution illustrated as a matrix product [21]

Some shenanigans with OVERLAP-SAVE

- The input FFT is computed from a **K-point sliding window of the input**. Before the transformation, the previous contents are shifted B samples to the left and the next length-B input block is stored rightmost.
- From the output of the K-point IFFT, the K−B leftmost samples are time-aliased and therefore discarded. The B rightmost samples are **saved into the output block.**
- Now we can just drop all the adding. The sliding window means we can just take blocks of samples from the ADC and write them out to the DAC in (hopefully) realtime and make sure latency is limited to our buffer size.

Overlap-SAVE partitions the OUTPUTS

$$\begin{bmatrix} y_0 \\ y_1 \\ y_2 \\ y_3 \\ \hline y_4 \\ y_5 \\ y_6 \\ y_7 \\ \hline y_8 \\ y_9 \\ y_{10} \\ y_{11} \end{bmatrix} = \left[\begin{array}{cccccccc} h_0 & 0 & 0 & 0 & 0 & 0 & 0 & 0 \\ h_1 & h_0 & 0 & 0 & 0 & 0 & 0 & 0 \\ h_2 & h_1 & h_0 & 0 & 0 & 0 & 0 & 0 \\ h_3 & h_2 & h_1 & h_0 & 0 & 0 & 0 & 0 \\ \hline h_4 & h_3 & h_2 & h_1 & h_0 & 0 & 0 & 0 \\ 0 & h_4 & h_3 & h_2 & h_1 & h_0 & 0 & 0 \\ 0 & 0 & h_4 & h_3 & h_2 & h_1 & h_0 & 0 \\ 0 & 0 & 0 & h_4 & h_3 & h_2 & h_1 & h_0 \\ \hline 0 & 0 & 0 & 0 & h_4 & h_3 & h_2 & h_1 \\ 0 & 0 & 0 & 0 & 0 & h_4 & h_3 & h_2 \\ 0 & 0 & 0 & 0 & 0 & 0 & h_4 & h_3 \\ 0 & 0 & 0 & 0 & 0 & 0 & 0 & h_4 \end{array}\right] \cdot \begin{bmatrix} x_0 \\ x_1 \\ x_2 \\ x_3 \\ x_4 \\ x_5 \\ x_6 \\ x_7 \end{bmatrix}$$

Figure 3.2.: Overlap-Save convolution as a matrix product [21]

You can implement these in MATLAB

- If you’re a masochist
- Probably better to use something nicer (and faster) like C/C++
- Seriously though, MATLAB is just good for proof of concept

PARTITIONED FAST CONVOLUTION

Kulp (1988) suggested segmenting RIR into short frames and store their FFTs (H1, H2, H3...)

- Input signal is processed in frames of the same length
- Can be implemented with 1 FFT and 1 IFFT per frame.

Convolution in HARDWARE

1999 - Sony DRE S777 - the first real-time convolution processor.

- Used samples of real spaces (impulse responses)
- Cost > $4k
- More realistic results than comb filters and allpass delay tricks

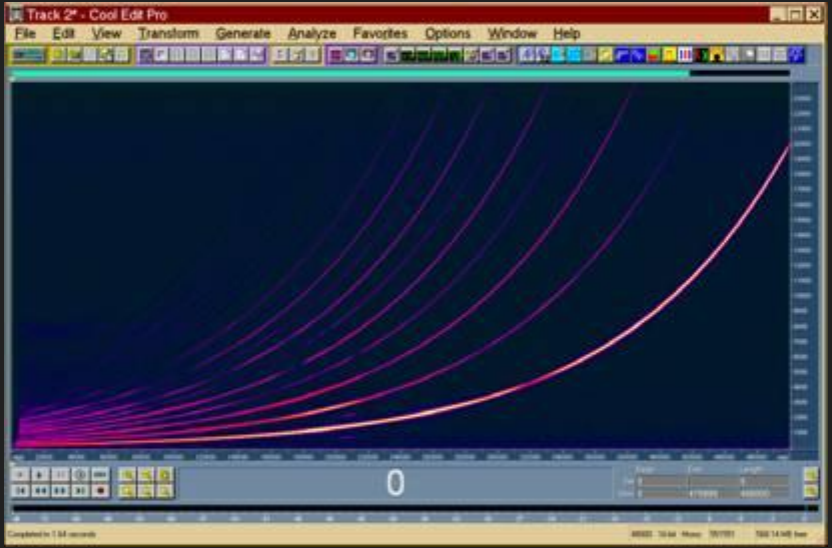
MODERN CONVOLUTION REVERB

- Efficiently computed using math libraries that utilize hardware optimizations - many instances can be running in real-time with little latency
- VST, AU, AAX plugin formats for use in DAWs
- Libraries of high-quality IRs using professional equipment and modern techniques
  - ESS (Exponential Sine Sweep) method (Angelo Farina)
  - Capture tools are automated and/or user-friendly!

ESS captures harmonic distortion

a pure sine sweep with constant amplitude and exponentially-increasing frequency

OUTPUT TEST SIGNAL



CAPTURED HARMONICS

MODERN CONVOLUTION REVERB

- Implemented in niche hardware units (Kemper Profiler / Axe-Fx III / Neural DSP Quad Cortex) that use preloaded IRs as well as user-defined capture memory.
    - IRs can be frequency responses that are piecewise components of a larger system - e.g., guitar cabinet speakers
- New methods to capture nonlinear responses such as Vacuum Tube based amplifiers
    - IR Switching - a different IR is employed depending on the amplitude of each sample of the signal to be filtered
    - Volterra kernels method (Farina)

Q&A Time

Recommended links:

https://ccrma.stanford.edu/~jos/pubs.html

http://pcfarina.eng.unipr.it/Public/Presentations/NonLinear_Convolution_Eng_files/frame.htm