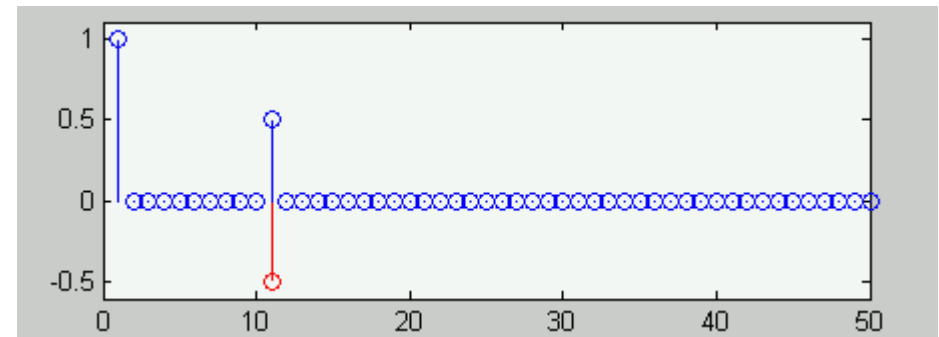
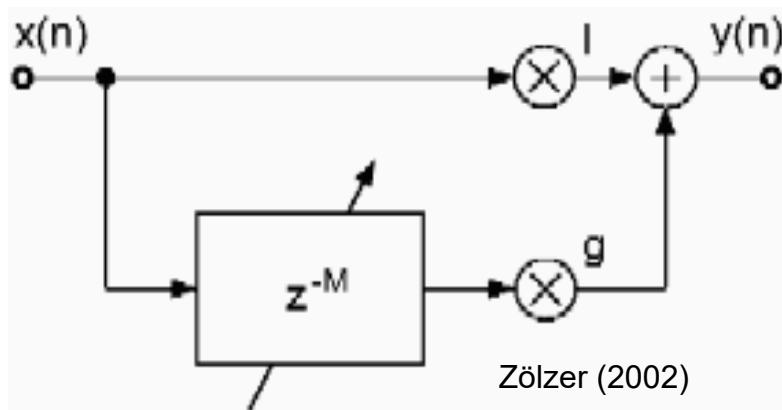


Allgemein

- Entstehung
- Unterteilung:
 - Nicht rekursive Strukturen
 - Rekursive Strukturen
- Auswirkungen
- Anwendungen:
 - Audioeffekte
 - Nachhallsimulation

FIR Kammfilter

- Struktur:

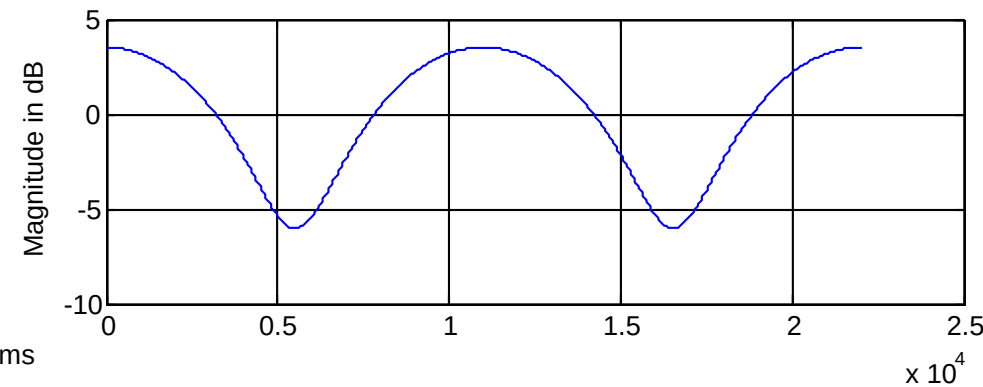
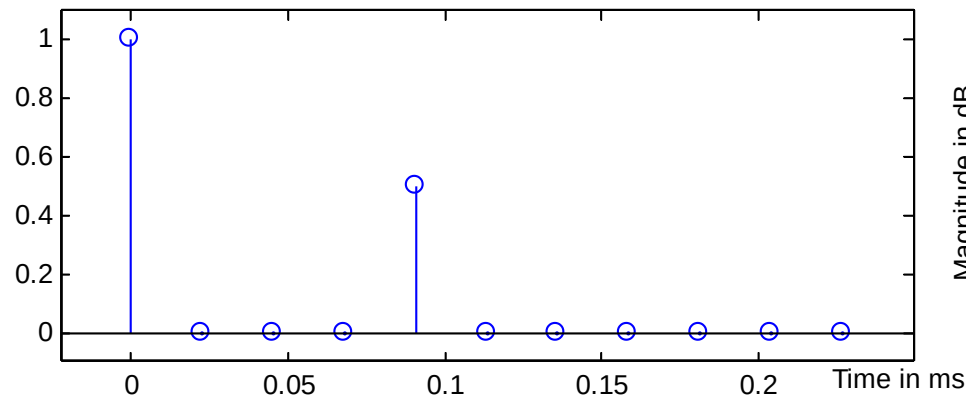


- Differenzengleichung: $y[n] = x[n] + g \cdot x[n - M]$

$$M = \tau \cdot f_s$$

- Übertragungsfunktion: $H(z) = 1 + g \cdot z^{-M}$

FIR Kammfilter: Parameter



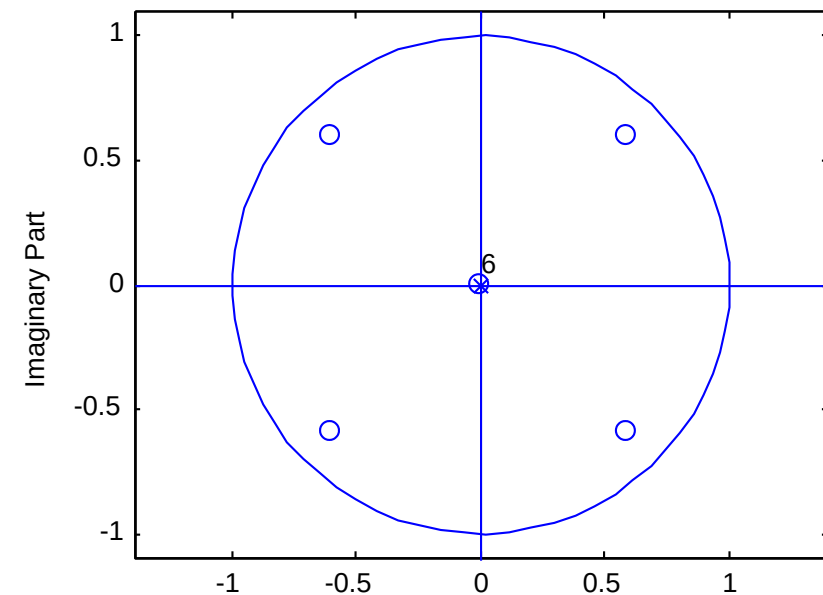
- für $g=1$: $|H(\omega)| = \sqrt{2[1 + \cos(\omega m)]}$

- für $g < 1$:

Peak: $P = 1 + g$

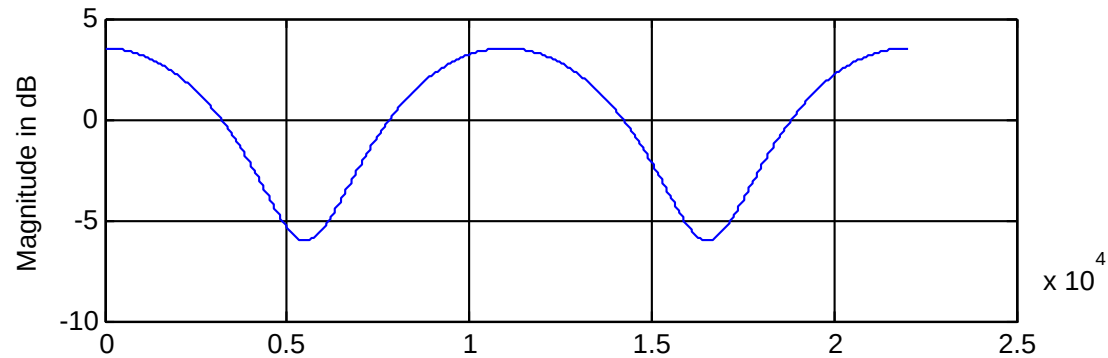
Valley: $V = 1 - g$

Peak-to-Valley ratio: $\frac{P}{V} = \frac{1+g}{1-g}$

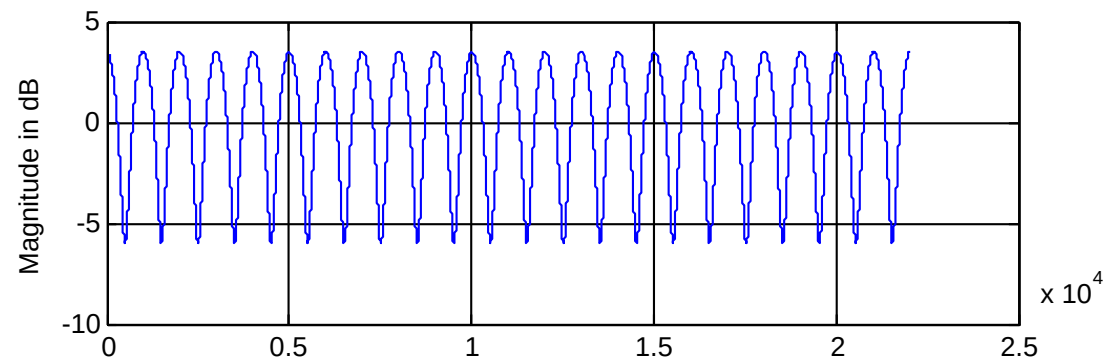


FIR Kammfilter: Vergleich

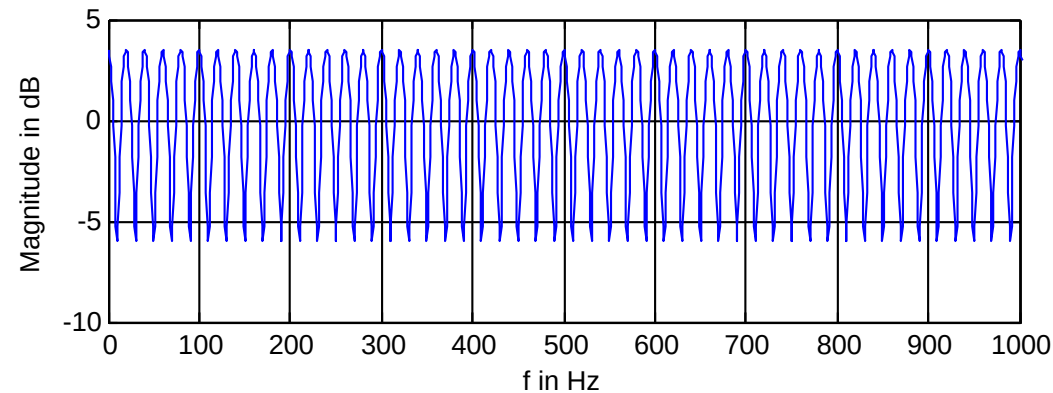
- 100 μ s



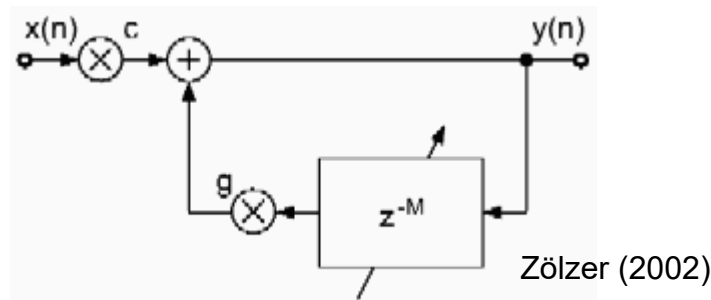
- 1ms



- 50ms

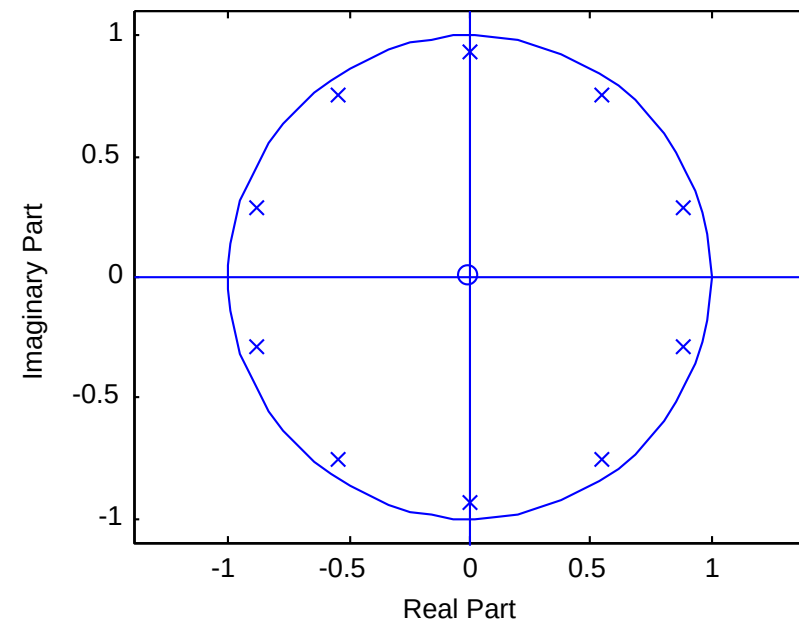
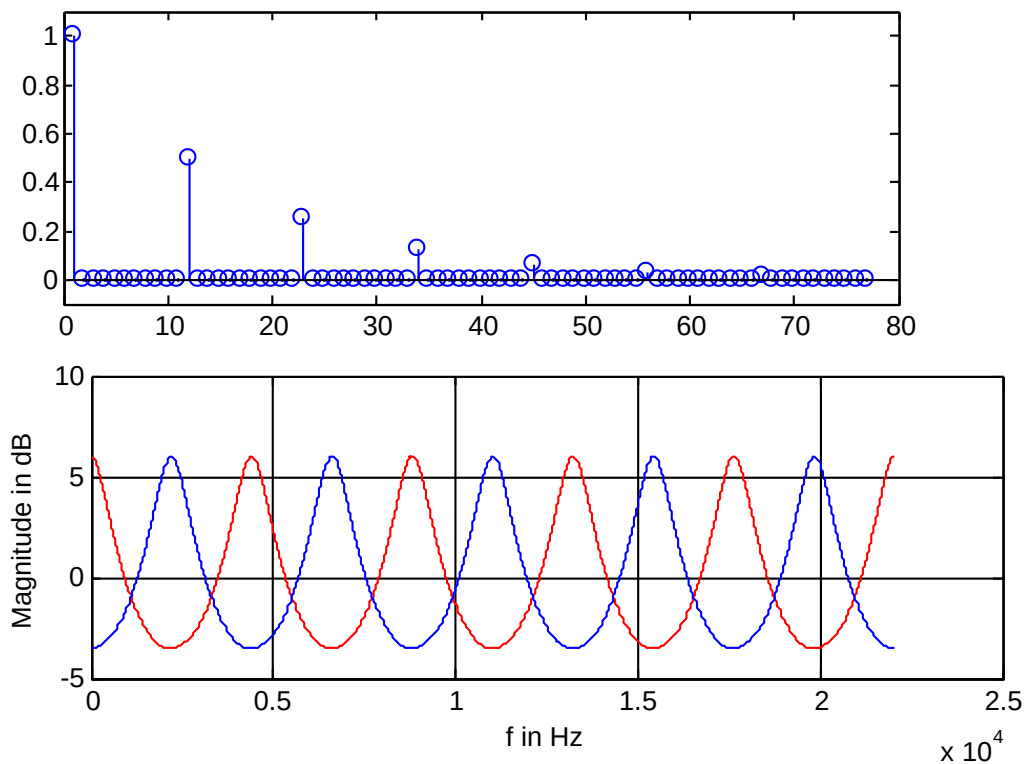


IIR Kammfilter



$$y[n] = c \cdot x[n] + g \cdot y[n - M] \quad M = \tau \cdot f_s$$

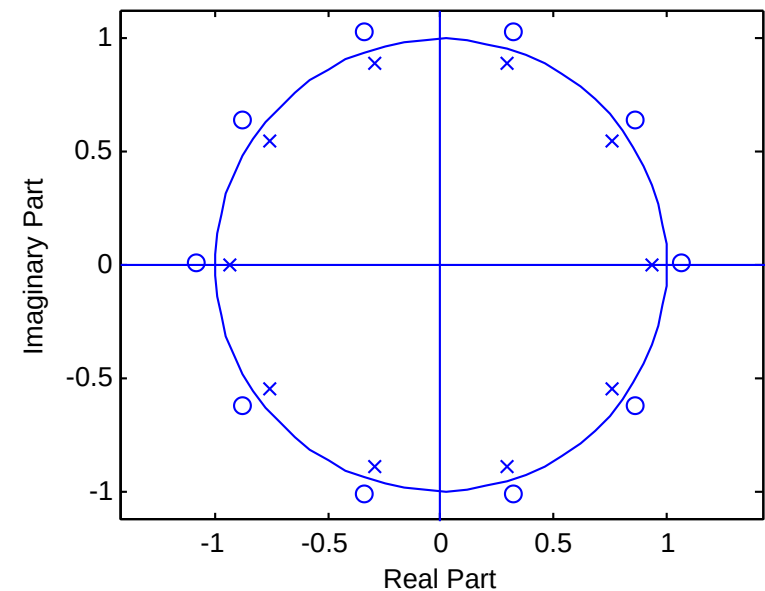
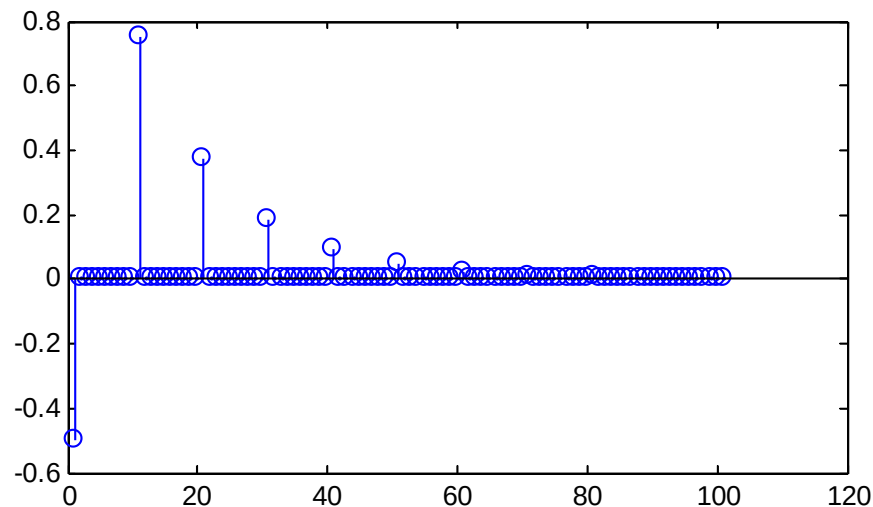
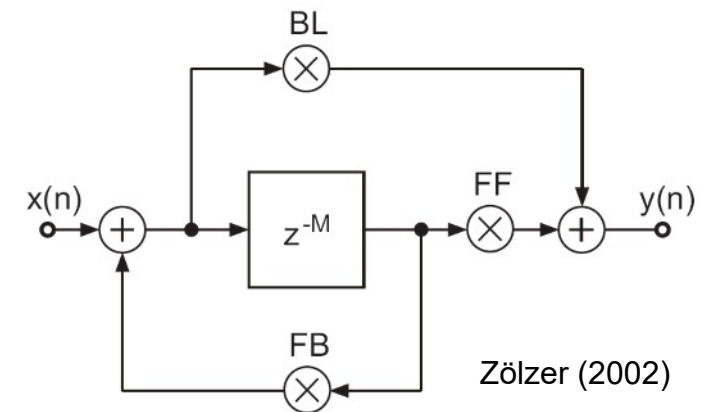
$$H(z) = \frac{c}{1 - g \cdot z^{-M}}$$



Allpass Kammfilter

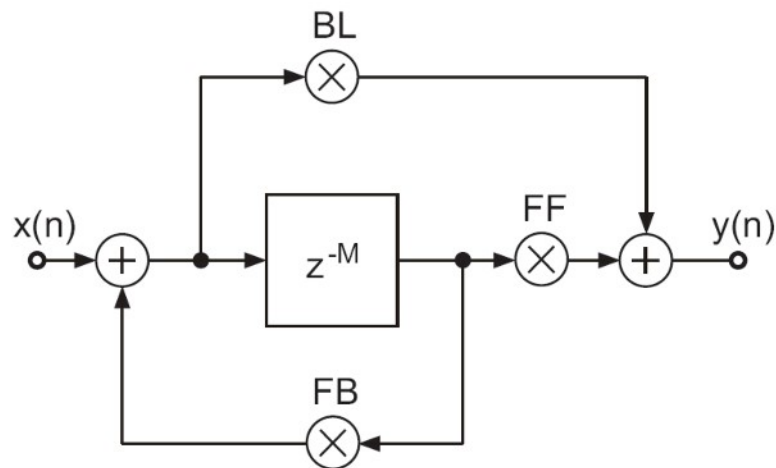
- Problem: Klangveränderung
- Lösung: Allpassfilter

$$H(z) = \frac{-B + z^{-M}}{1 - B \cdot z^{-M}} \quad |H(z)| = 1$$



Allgemeine Kammfilterstruktur

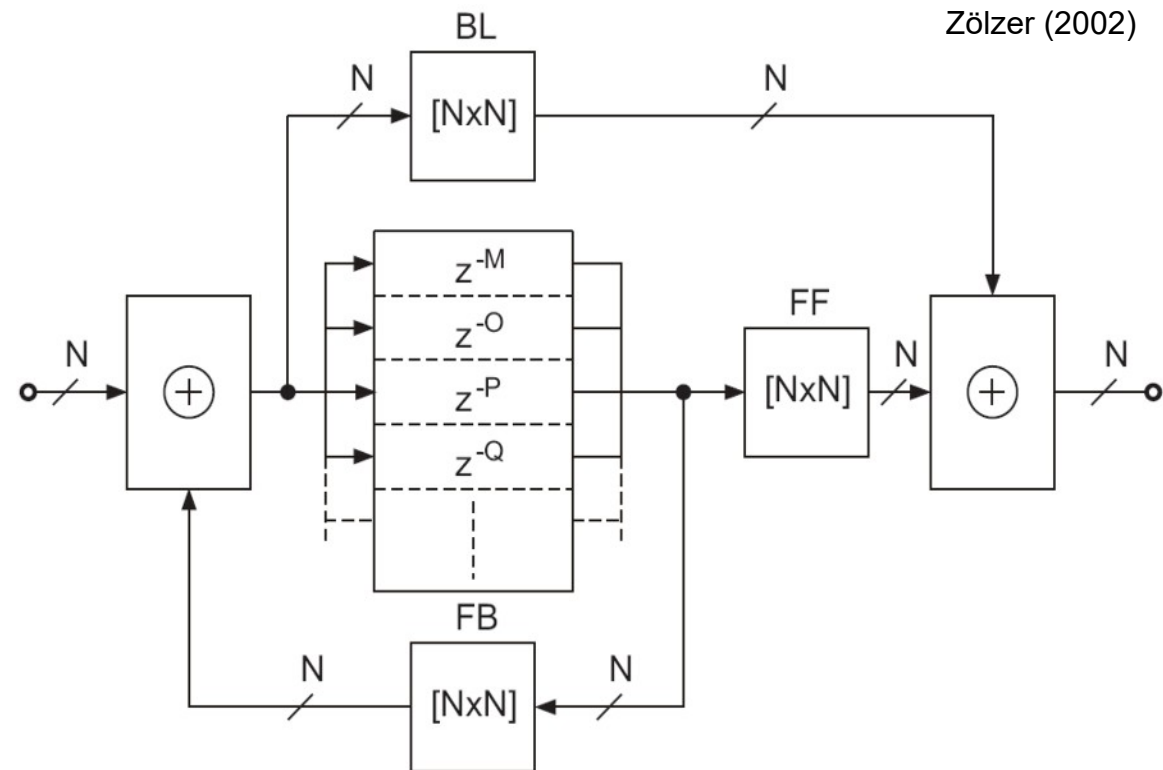
- Einfach:



Zölzer (2002)

	BL	FB	FF
FIR comb filter	X	0	X
IIR comb filter	1	X	0
allpass	a	$-a$	1
delay	0	0	1

- Matrix:



Zölzer (2002)

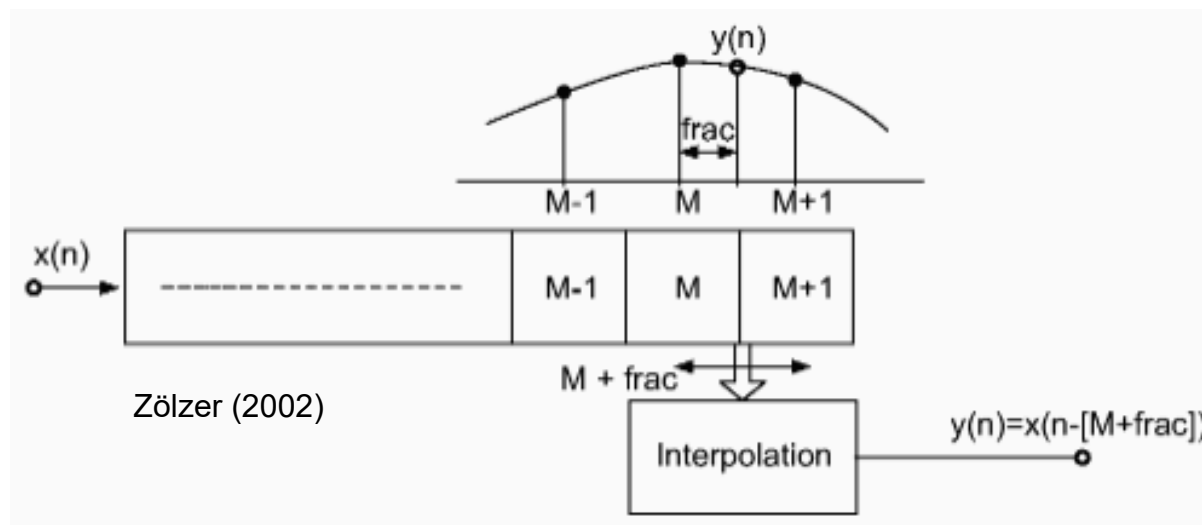
	delay	BL	FB	FF
slapback	50ms	1	0	X
echo	>50ms	1	$0 < X < 1$	0
reverb		matrix	matrix	matrix

Fraktionelle Verzögerungsglieder

- Problem: vorgegebene Verzögerung des Signals wird durch Diskretisierung quantisiert:
 - Zeitliche Diskretisierung (bei $f_s = 44.1$ kHz): $22.7 \mu\text{s}$
 - Räumliche Diskretisierung (bei Raumtemperatur): 7.5 mm
- Ansätze:
 - Höhere Abtastrate wählen
 - Fraktionelle Verzögerungsalgorithmen verwenden

Fraktionale Delays: Prinzip

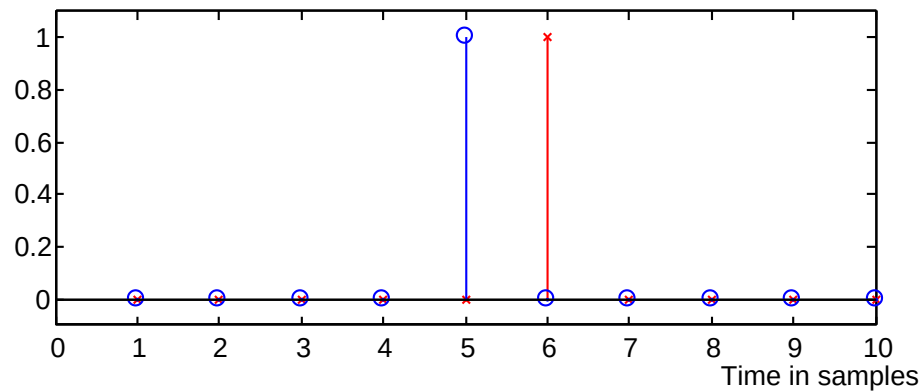
- Ziel: $y[n] = x[n - (M \cdot frac)]$



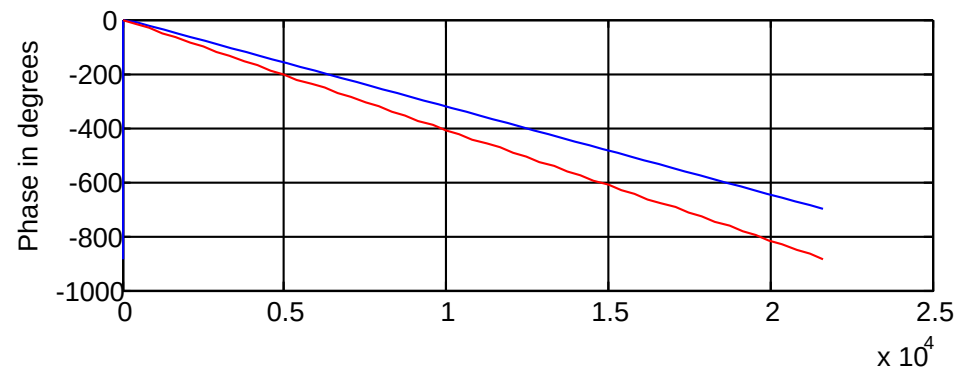
- Aufgabe trennen: $M \cdot frac = M + frac$
 $y[n] = x[n - (M + frac)]$

Fraktionelle Delays: Ansatz

- Zeitbereich:

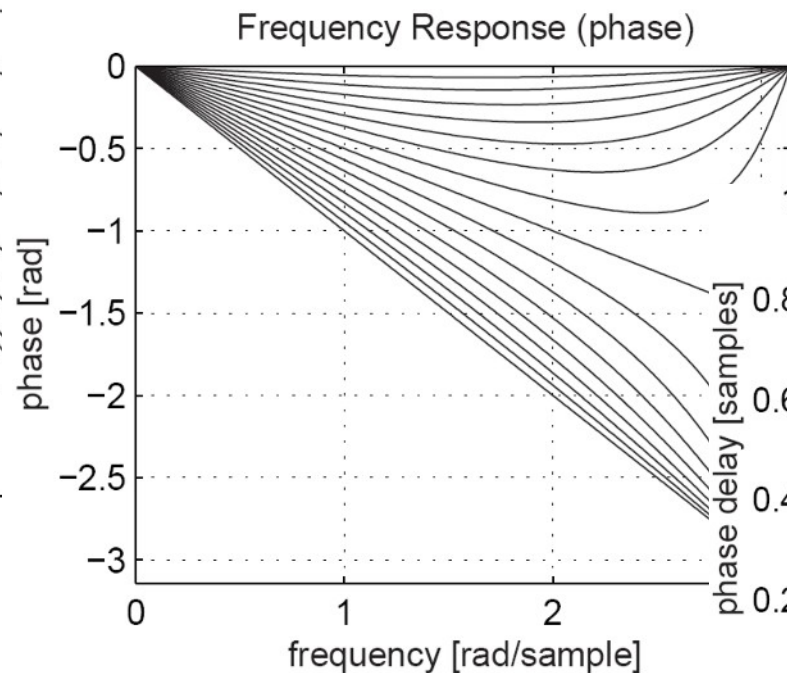
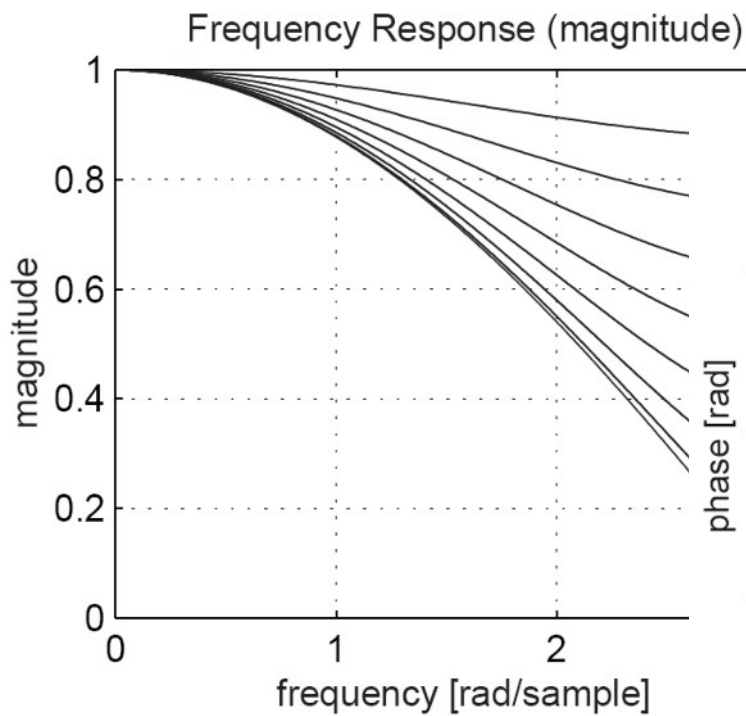


- Frequenzbereich:



FIR Interpolationsfilter

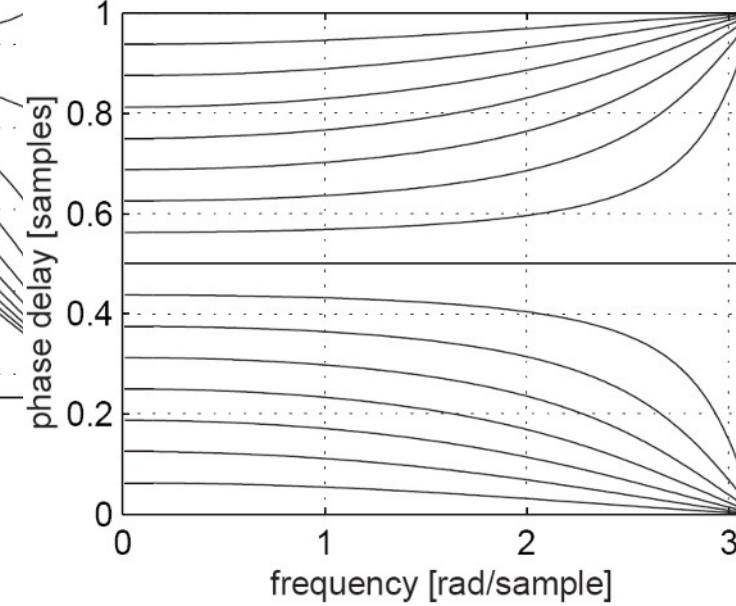
- Interpolation 1. Ordnung: $H(Z) = c_0 + c_1 z^{-1}$
- Spezialfall Mittelwert: $c_0 = 0.5$ und $c_1 = 0.5$



Rochesso (2003)

$$c_0 + c_1 = 1$$

$$c_1 \approx \tau_{ph}$$

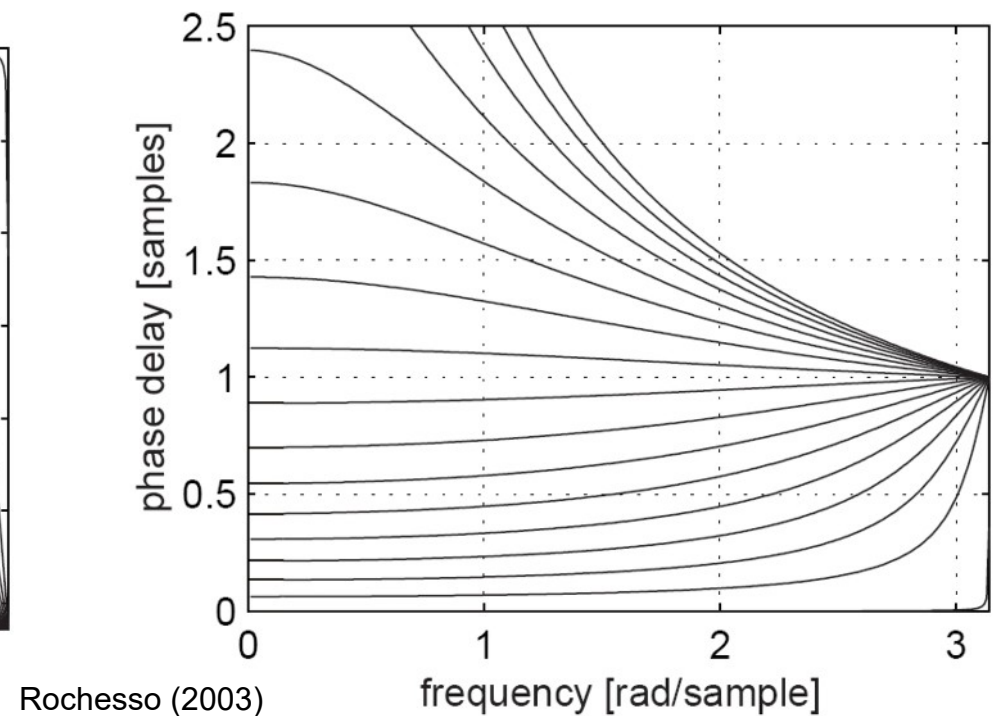
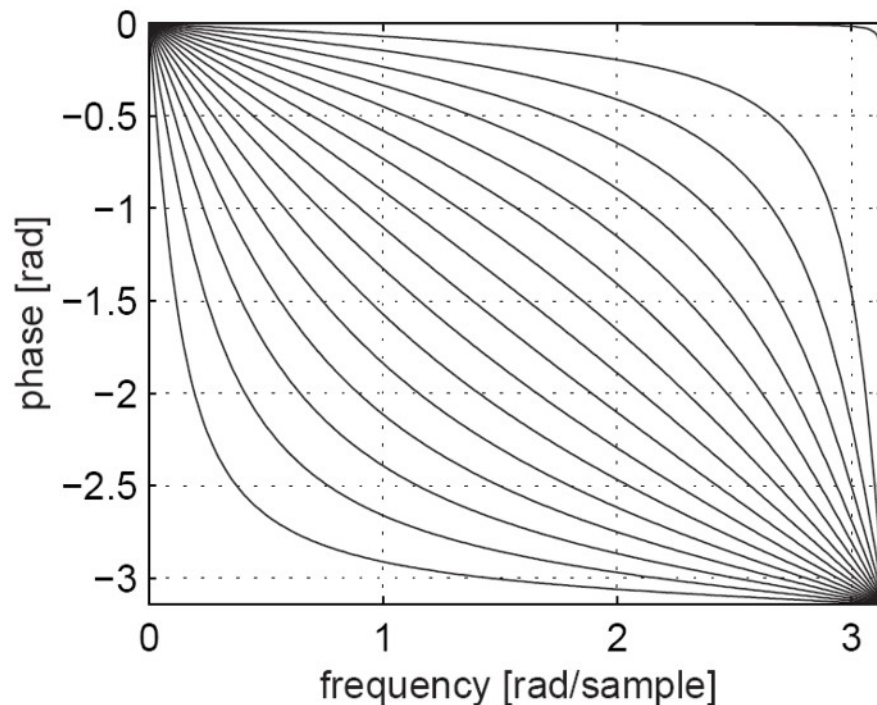


Allpass Interpolationsfilter

- Flacher Amplitudengang? Allpass:

$$H(z) = \frac{-c + z^{-m}}{1 - c \cdot z^{-M}}$$

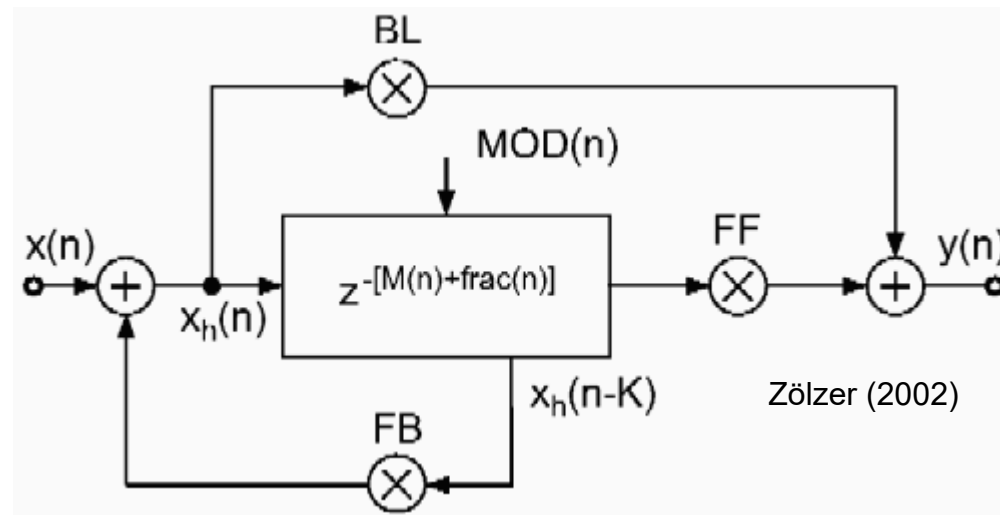
$$c \approx \frac{1 - \tau_{ph}(0)}{1 + \tau_{ph}(0)}$$



Rochesso (2003)

Verzögerungsglieder

- Allgemeine Struktur:

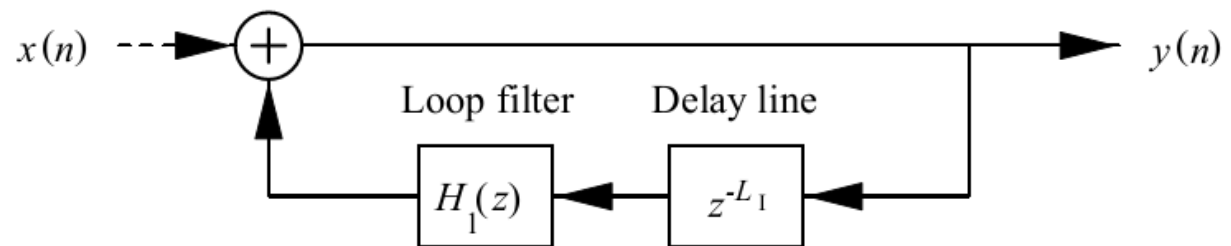


<i>Effekt</i>	<i>BL</i>	<i>FF</i>	<i>FB</i>	<i>Delay [ms]</i>	<i>MOD(n)</i>
Resonator	-	$0 < x < 1$	-	0..20	-
Slapback	-	$0 < x < 1$	-	20..50	-
Echo	-	$0 < x < 1$	-	> 50	-
Vibrato	-	1	-	0	0.1-5 Hz, sinus
Flanger	0.7	0.7	0.7	0	0.1-1 Hz, sinus
Chorus	0.7	1	-	2..30	LP-Rauschen
Doubling	0.7	0.7	-	30..100	LP-Rauschen

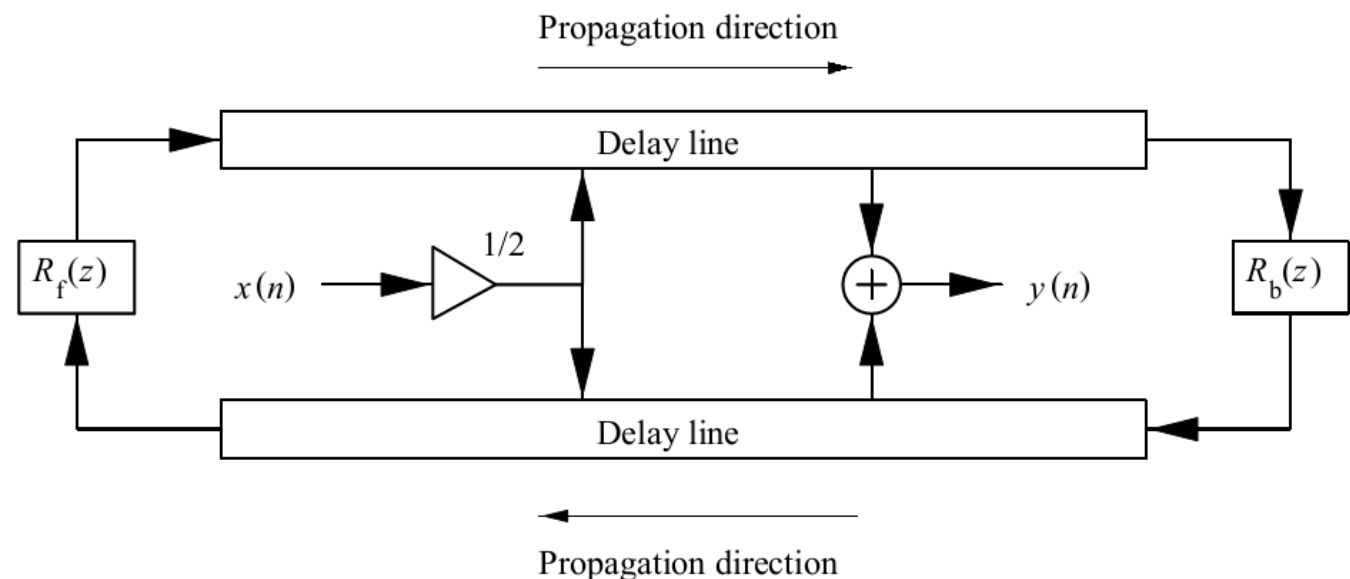


Spezielle Strukturen: Simulation vibrierender Saiten

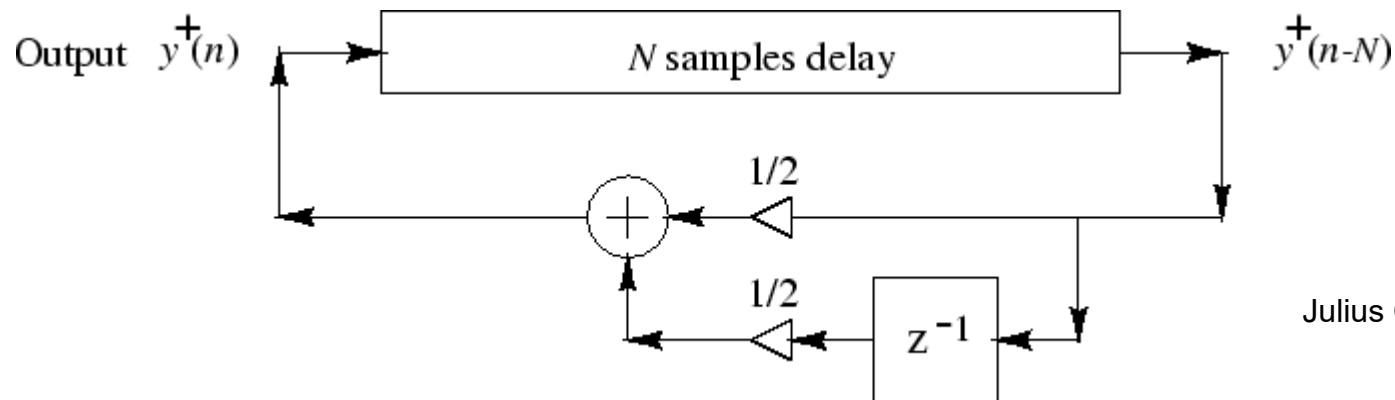
- Karplus und Strong (1983):



- Smith (1987):



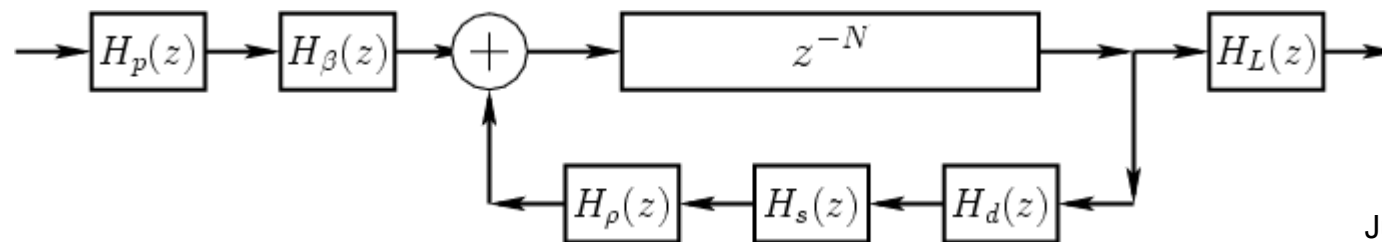
Karplus-Strong Algorithmus



Julius O. Smith III (2006)

- Modell einer vibrierenden Saite
 - Gedämpfte Saite mit frequenzabhängigen Dämpfung
 - Interpretierbar als:
 - Vereinfachtes Wave-Guide-Model
 - Rekursiver Kammfilter mit LP-Filter in der Rückkopplung

Extended Karplus-Strong Algorithmus



Julius O. Smith III (2006)

N = pitch period ($2 \times$ string length) in samples

$$H_p(z) = \frac{1-p}{1-pz^{-1}} = \text{pick-direction lowpass filter}$$

$$H_\beta(z) = 1 - z^{-\beta N} = \text{pick-position comb filter, } \beta \in (0, 1)$$

$$H_d(z) = \text{string-damping filter (one/two poles/zeros typical)}$$

$$H_s(z) = \text{string-stiffness allpass filter (several poles and zeros)}$$

$$H_\rho(z) = \frac{\rho(N) - z^{-1}}{1 - \rho(N)z^{-1}} = \text{first-order string-tuning allpass filter}$$

$$H_L(z) = \frac{1 - R_L}{1 - R_L z^{-1}} = \text{dynamic-level lowpass filter}$$

Weiterführende Literatur

- Davide Rocchesso. Introduction to Sound Processing.
<http://profs.sci.univr.it/~rocchess/SP>
- T. I. Laakso, V. Välimäki, M. Karjalainen, and U. K. Laine. (1996) “Splitting the Unit Delay — Tools for Fractional Delay Filter Design,” IEEE Signal Processing Magazine, 13: 30–60.
- J. Dattorro. (1997). “Effect design - part 2: Delay-line modulation and chorus,” J. Audio Eng. Soc., 45: 764 – 788.
- M. Karjalainen, V. Välimäki, and T. Tolonen (1998). “Plucked-String Models, from the Karplus-Strong Algorithm to Digital Waveguides and Beyond,” Computer Music Journal, 22: 17 – 32.
- Smith III, J. O. (2006) “Elementary Digital Waveguide Models for Vibrating Strings”, MUS420/EE367A Lecture 5A, Stanford University.