RECENT INTERESTING AND USEFUL ENHANCEMENTS OF POLYPHASE FILTER BANKS

Parts of presentations made at the
23rd International Symposium on
Wireless Personal Multimedia Communications
(WPMC 2020 Virtual Edition)

18-21 October 2020

Okayama, Japan

Best Paper Award at Conference

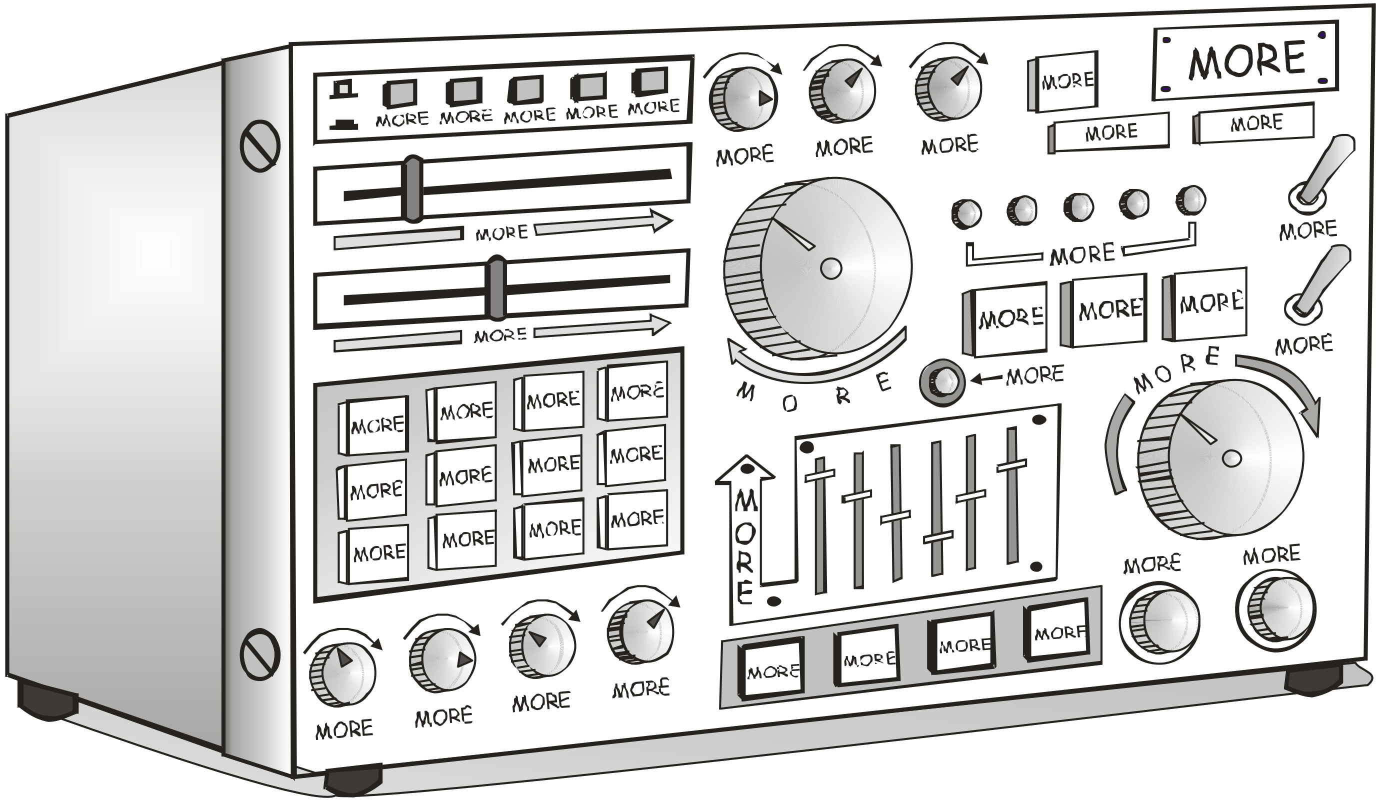
fred harris

UC San Diego
Jacobs School of Engineering

Atlanta Section Chapter, COM19
Buenaventura Section Chapter, COM19
Coastal Los Angeles Sect Chapter, SP01, VT06/Com19
Seattle Sect Jt. Chapter, COM19/VT06/BT02/IT12/ITS38

IEEE
Advancing Technology for Humanity

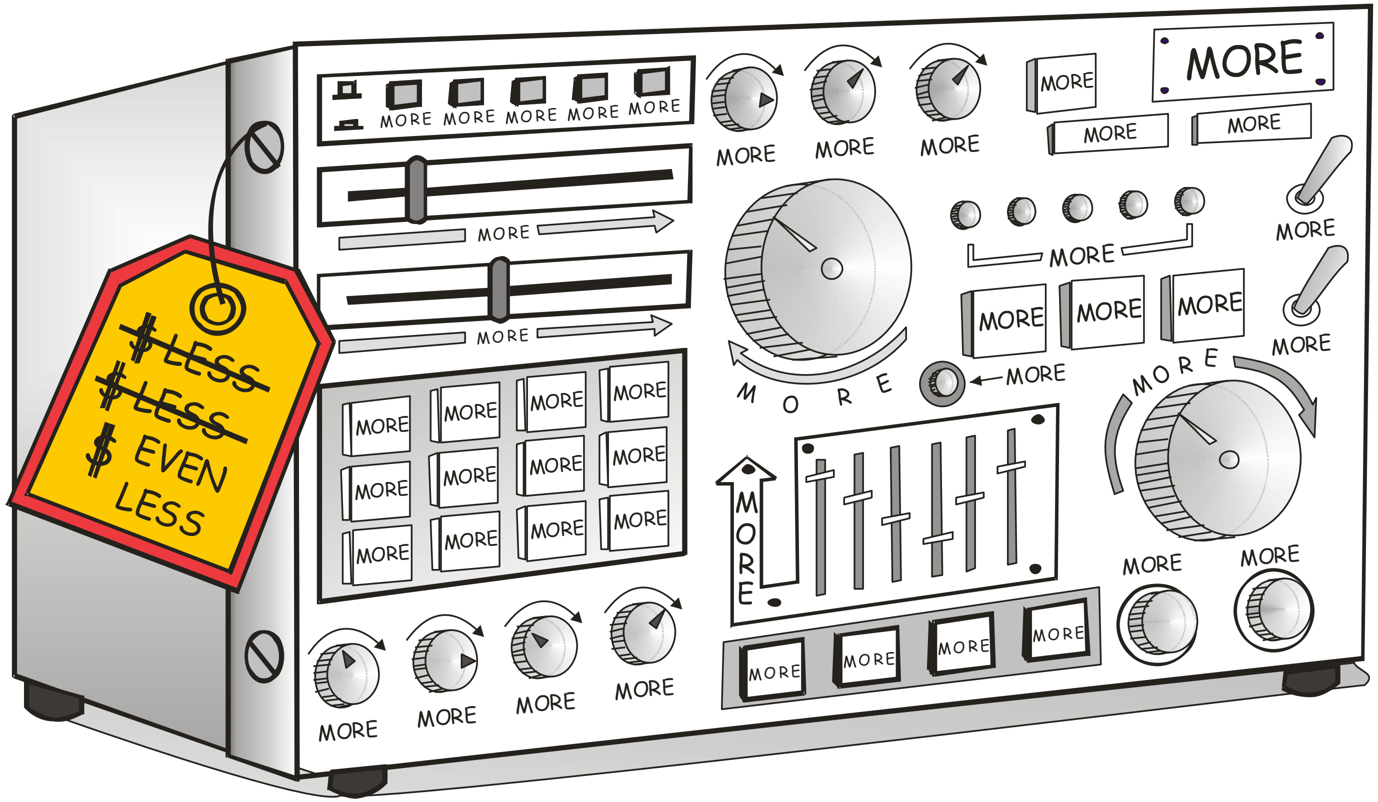
What the Customer Wants



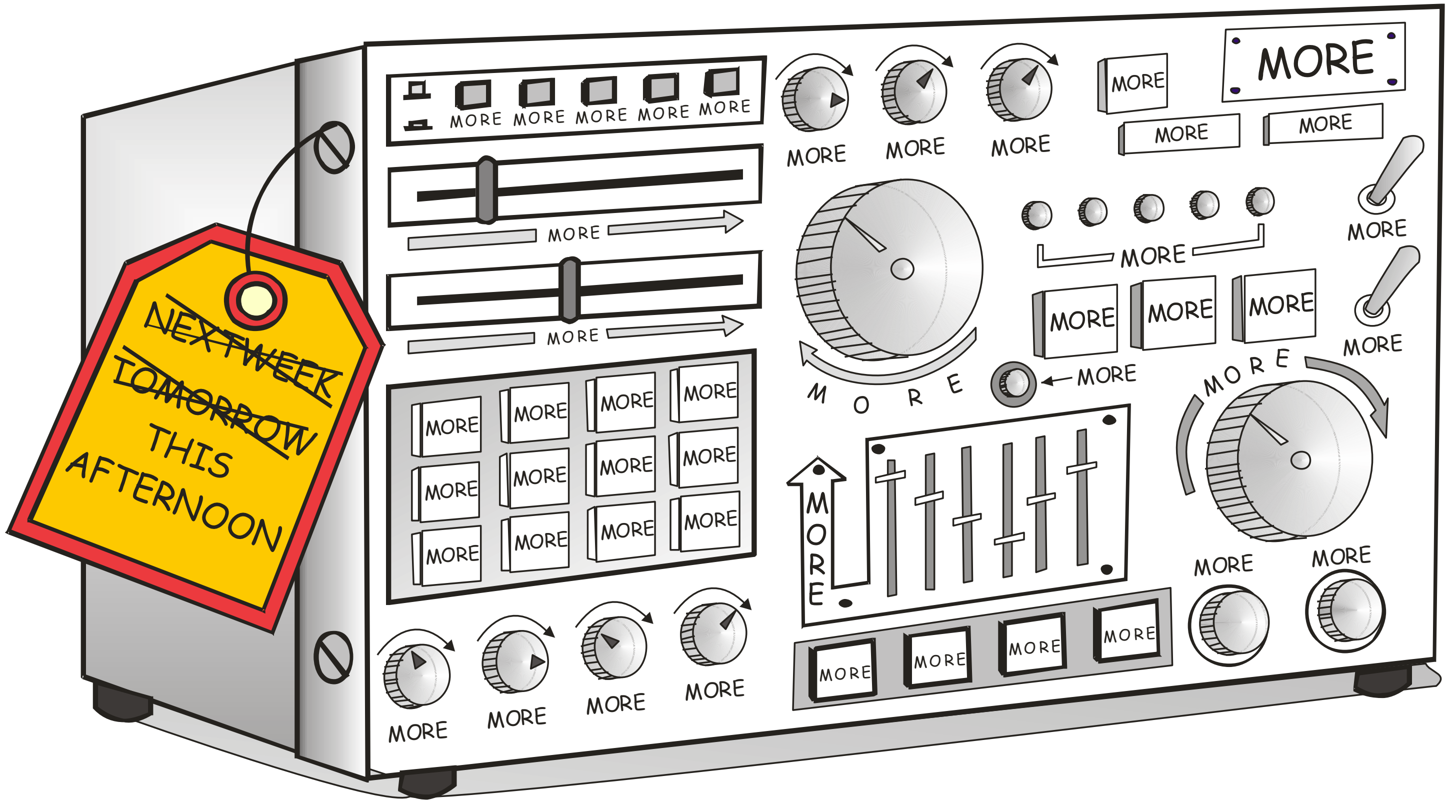
What ever you told the Customer, this is what he/she Heard:

More Bandwidth
More Dynamic Range
More Formats
More Options
More Bells
More Whistles
More
More
More
More

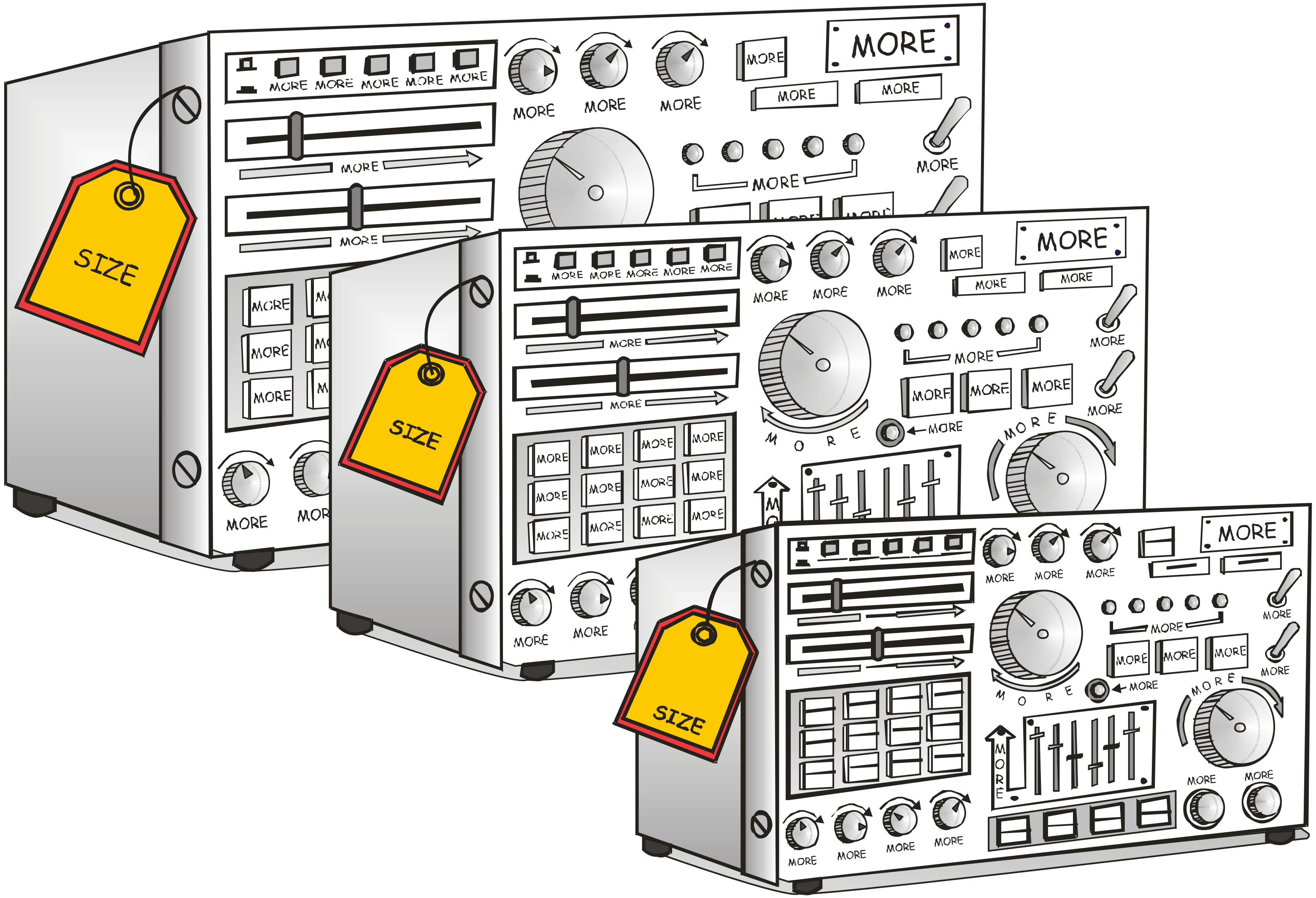
What the Customer Expects to Pay



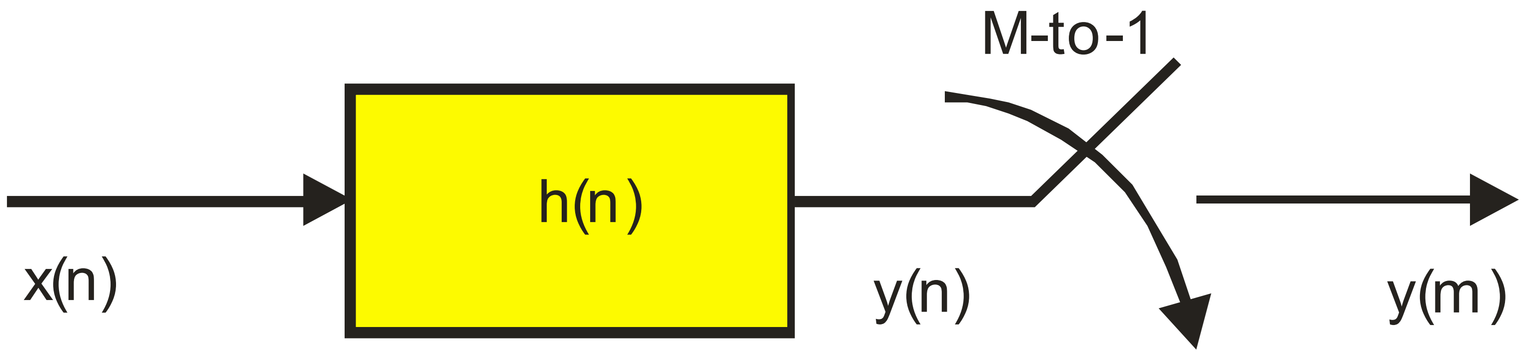
When the Customer Wants it.



Size the Customer Wants.



When a sampled data filter reduces bandwidth, it makes sense that the system should also reduce sample rate.
It would be interesting if the filter could do both; reduce BW and reduce sample rate

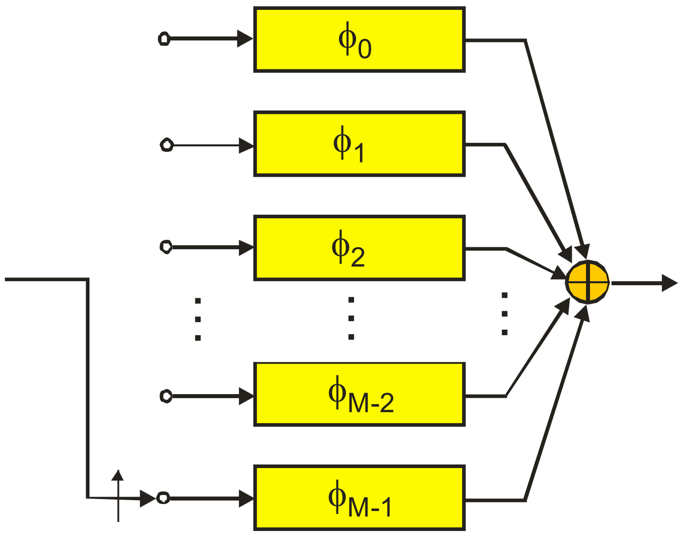


Re-Sampling!

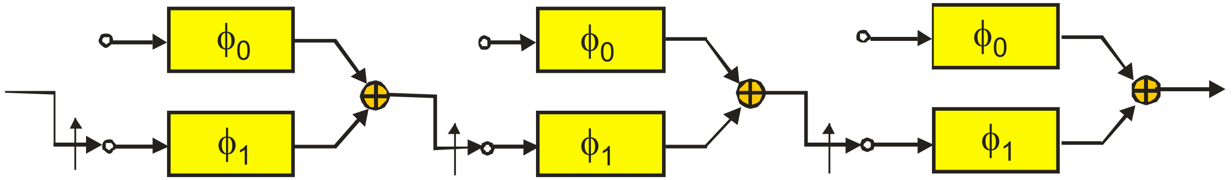
Does That Mean
We Didn't Do it Right
the First time?

Common Down Sampling Filters

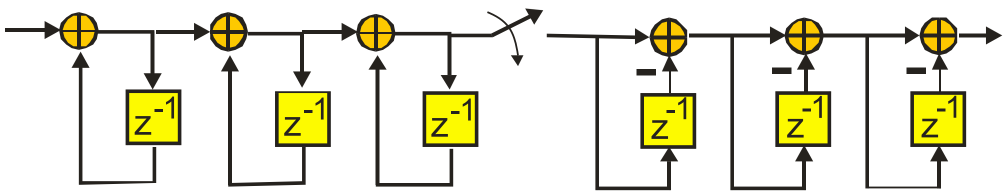
M-samples in
1-sample out



POLYPHASE FILTER

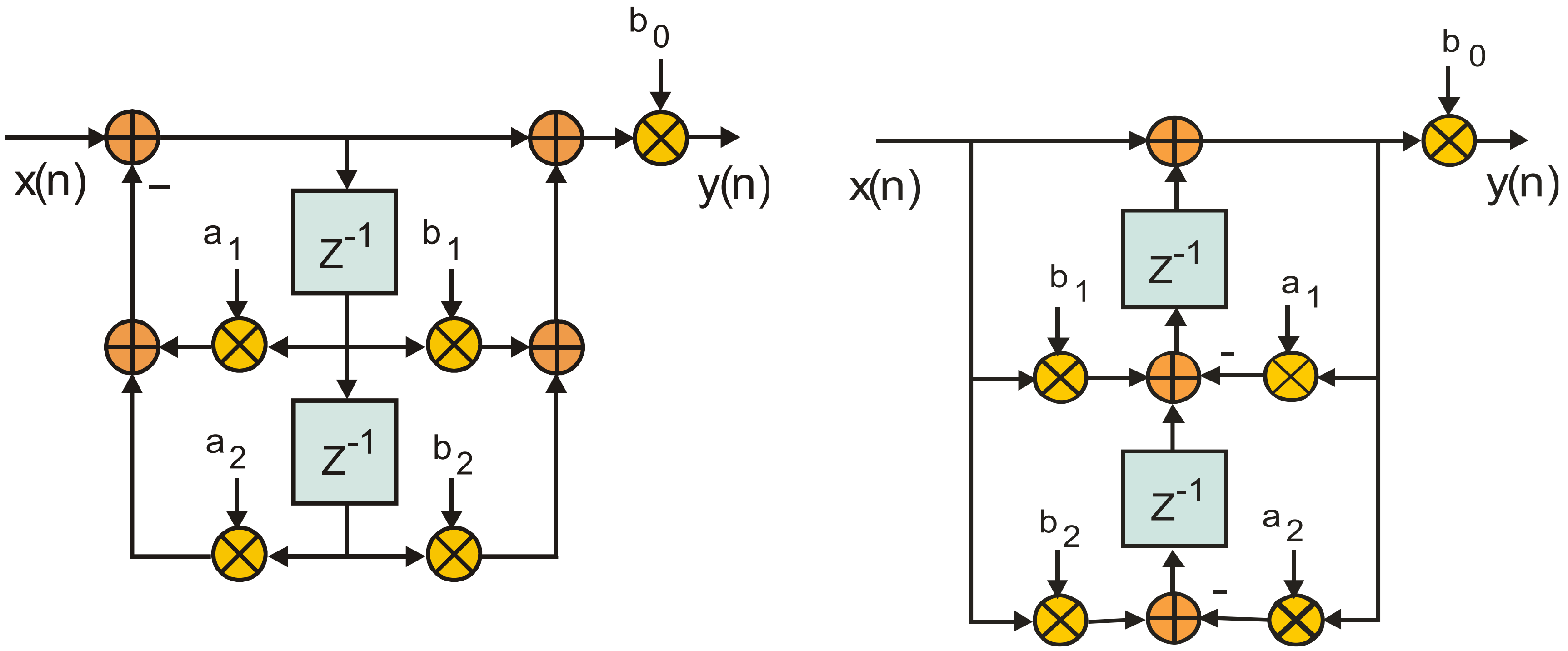


DYADIC HALF-BAND FILTER



HOGENAUER FILTER (CIC)

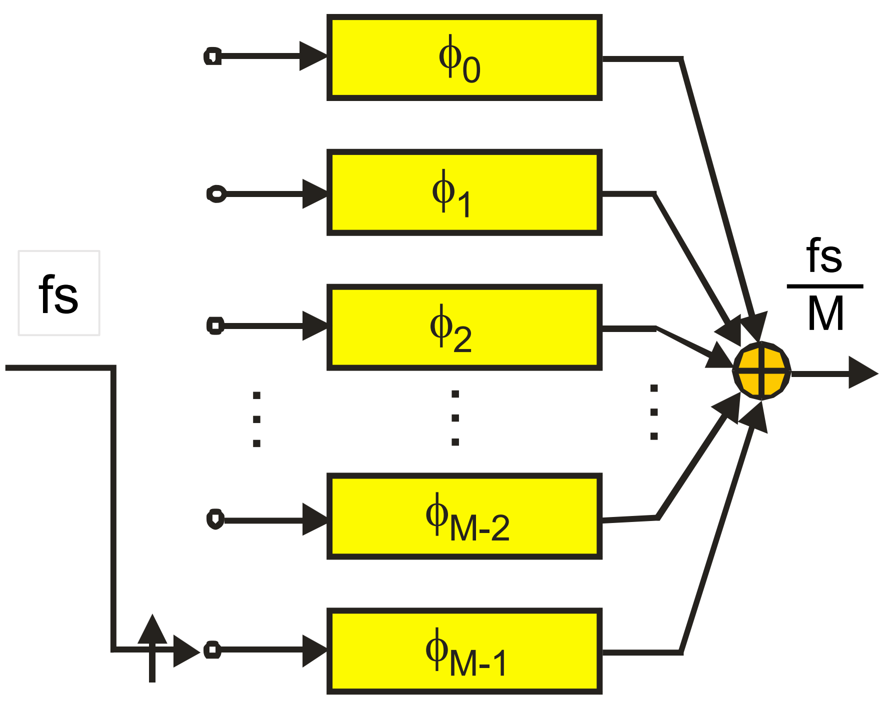
Dual LTI Filters Have Same Transfer Function



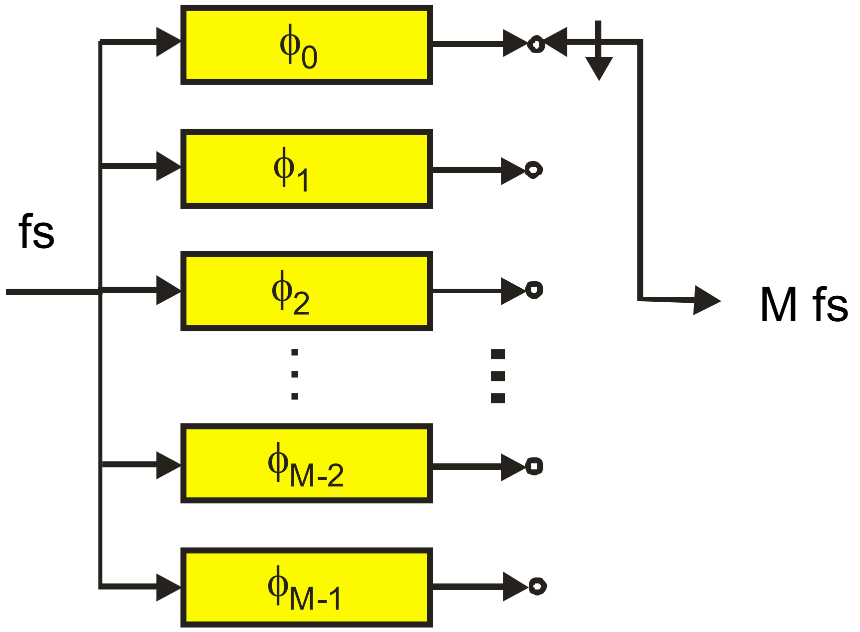
Dual Graphs:
Replace Nodes with Summing Junctions,
Replace Summing Junctions with Nodes,
Reverse Direction of Arrows.

Dual graph is
Seen as transposing
State transition matrix

Dual LTV Filters Perform Opposite Function



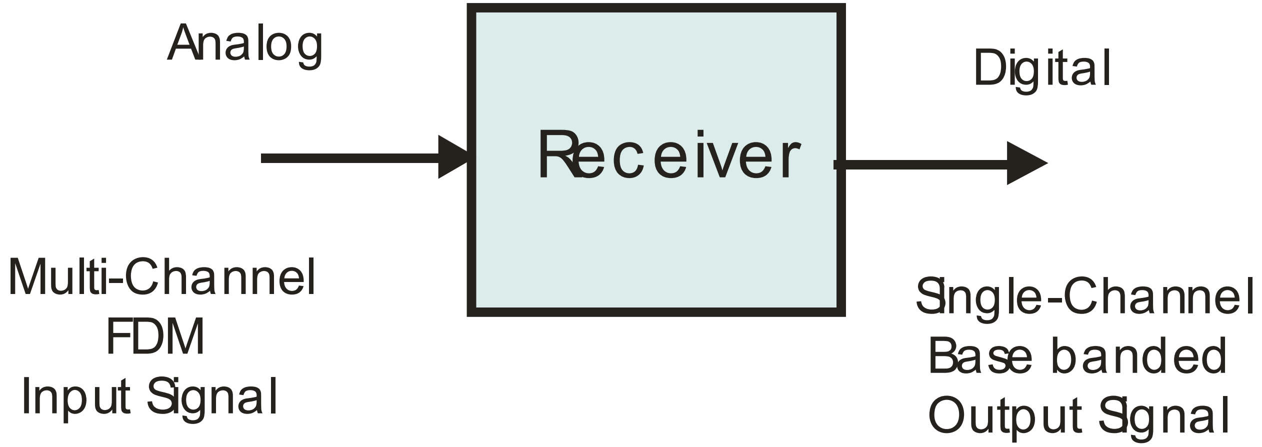
M-to-1 Down Sampler



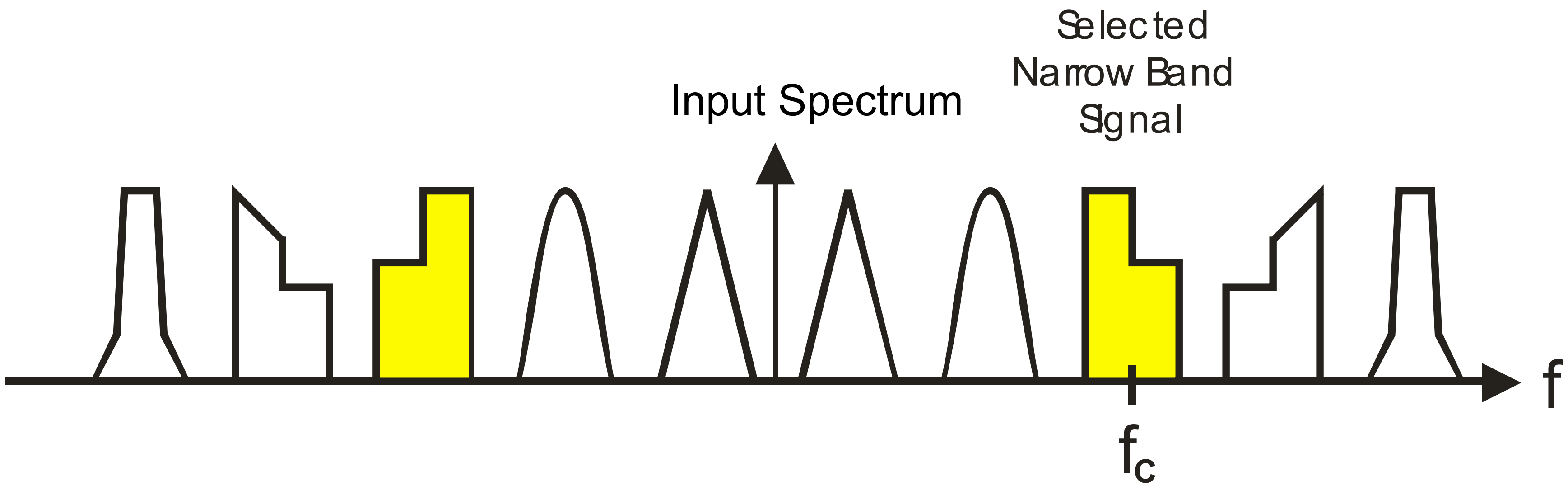
1-to-M Up Sampler

Motivation For Using Multirate Filters

Processing Task: Obtain Digital Samples of Complex Envelope Residing at Frequency f_C



Applause from Audience
When I presented this slide
At Signal Processing Society
Meeting at
Brigham Young University
Michael Rice was my host



See!

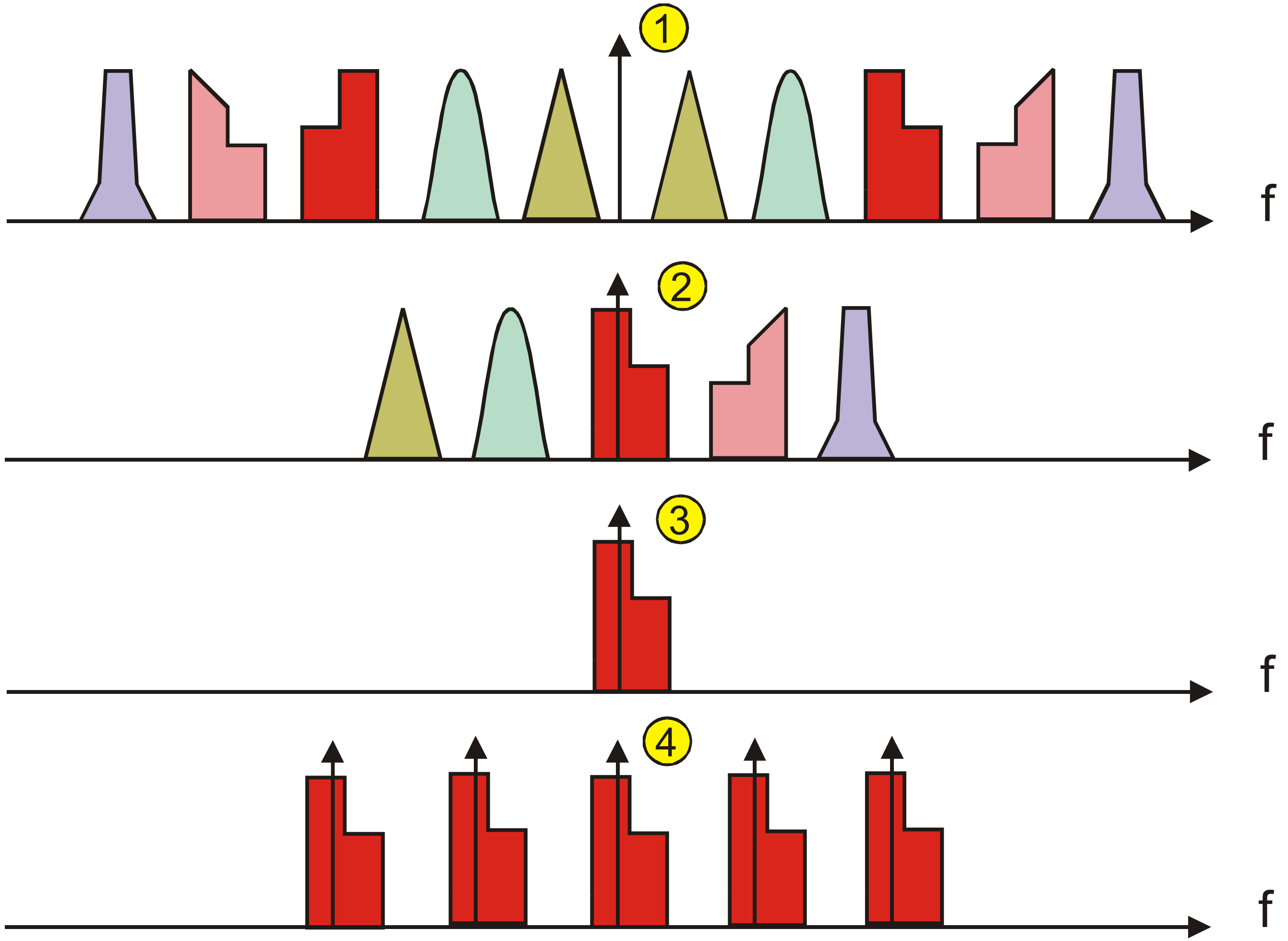

Washington
Montana
North Dakota
Minnesota
Maine
Oregon
Idaho
Wyoming
South Dakota
Wisconsin
Michigan
Vt.
N.H.
Mass.
New York
Conn.
R.I.
Nevada
Utah
Nebraska
Iowa
Pennsylvania
New Jersey
Illinois
Indiana
Ohio
Md.
Delaware
Colorado
Kansas
West Virginia
Virginia
D.C.
California
Missouri
Kentucky
North Carolina
Tennesee
Oklahoma
Arizona
New Mexico
Arkansas
South Carolina
Miss
Alabama
Georgia
Texas
Louisiana
Alaska
Hawaii
Florida

DSP Insertion in Communication Systems

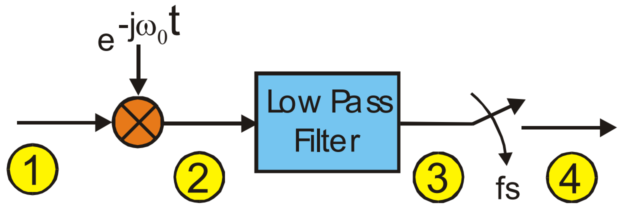
Instinctive First Response:
Copy Legacy Analog Prototype

- We should avoid this approach!!
- If we don't, we emulate an analog design!
- That is not the reason we invoke and apply DSP!
- DSP is inserted to improve performance and reduce cost!
- Analog prototype systems incorporate design compromises appropriate for the time they were made!
- We don't want to perpetuate those compromises!
- We have access to tools and resources not available to past designers!

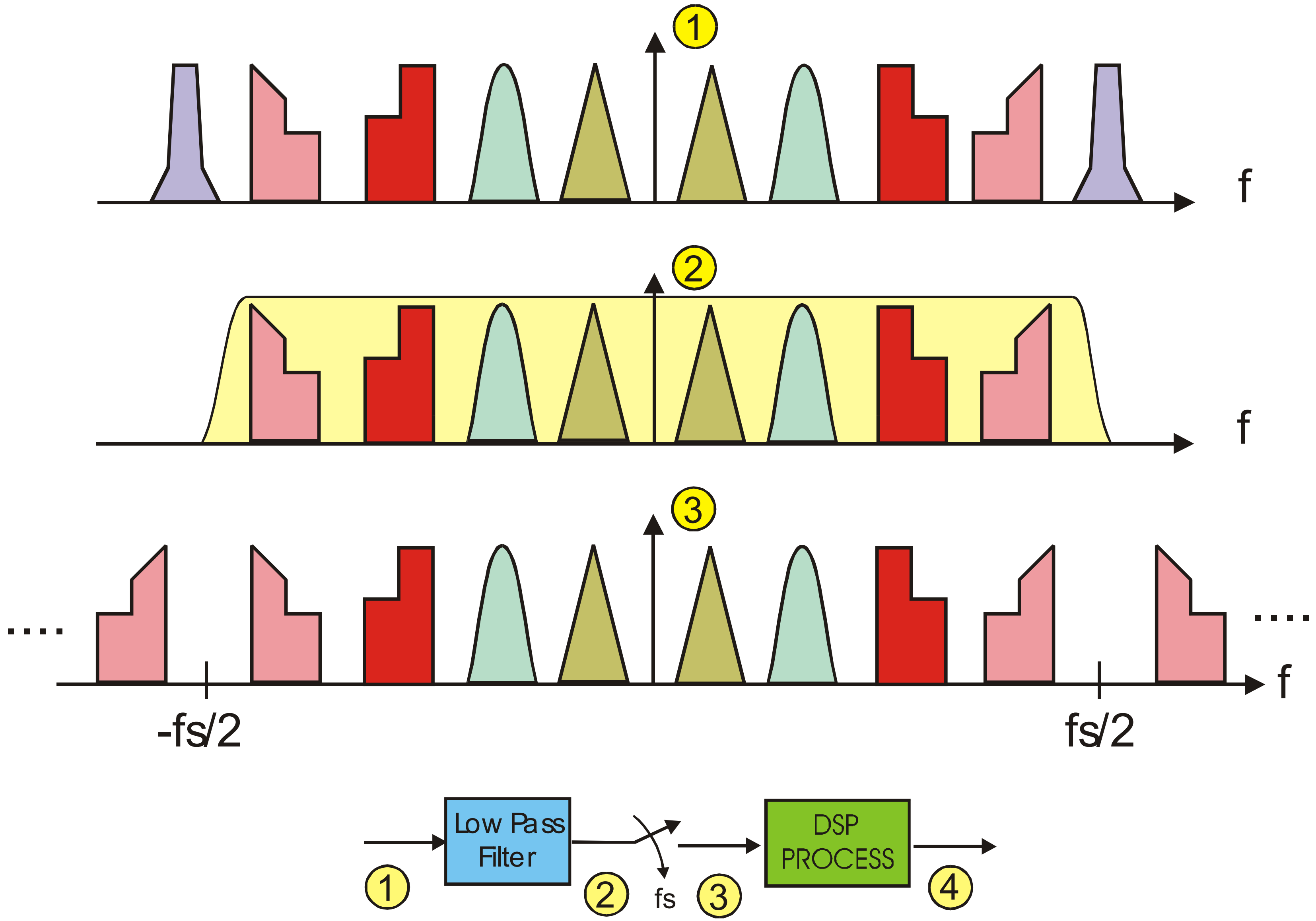
First Generation DSP Receiver



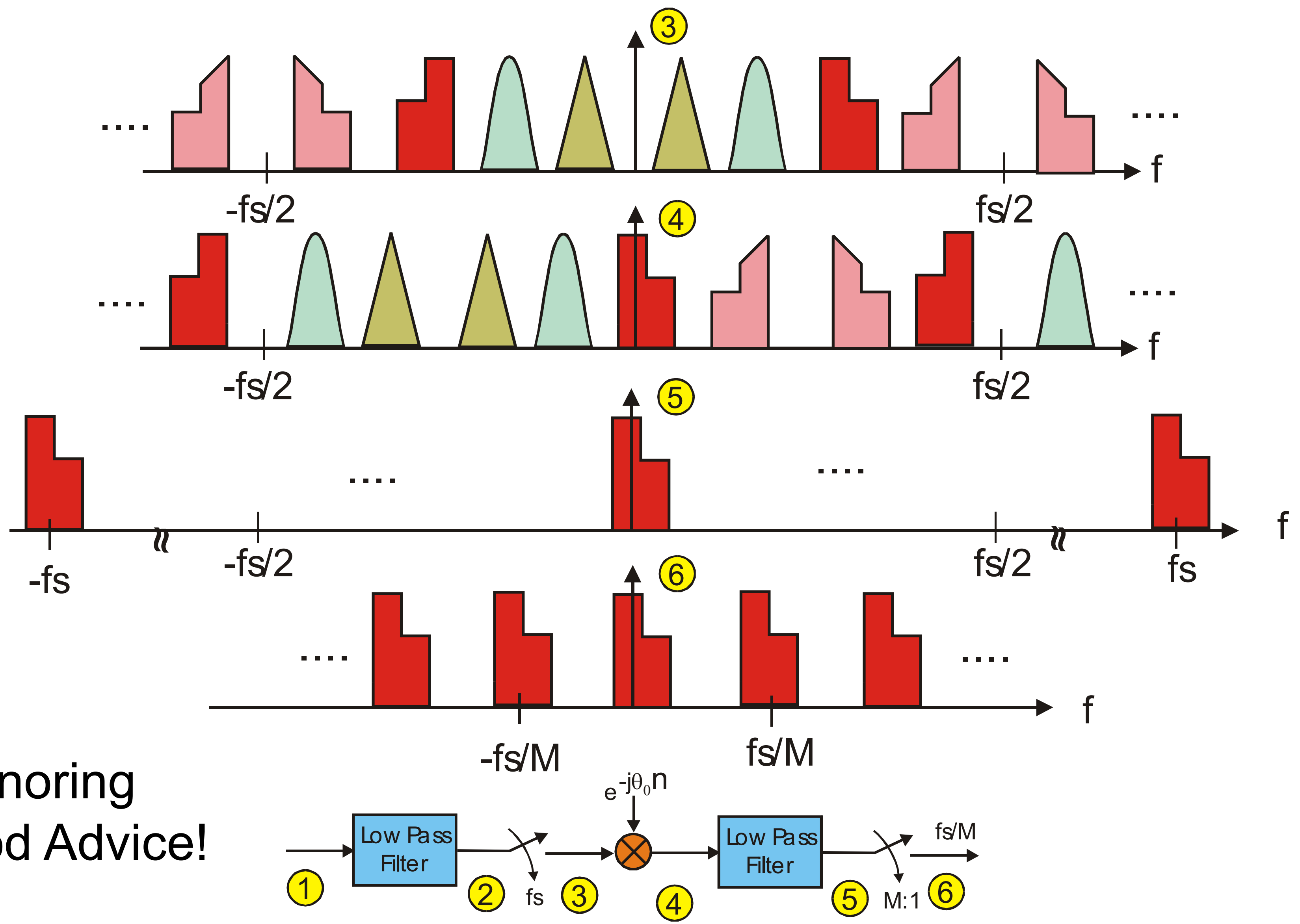
Analog
Signal Processing



Signal Conditioning for DSP Receiver

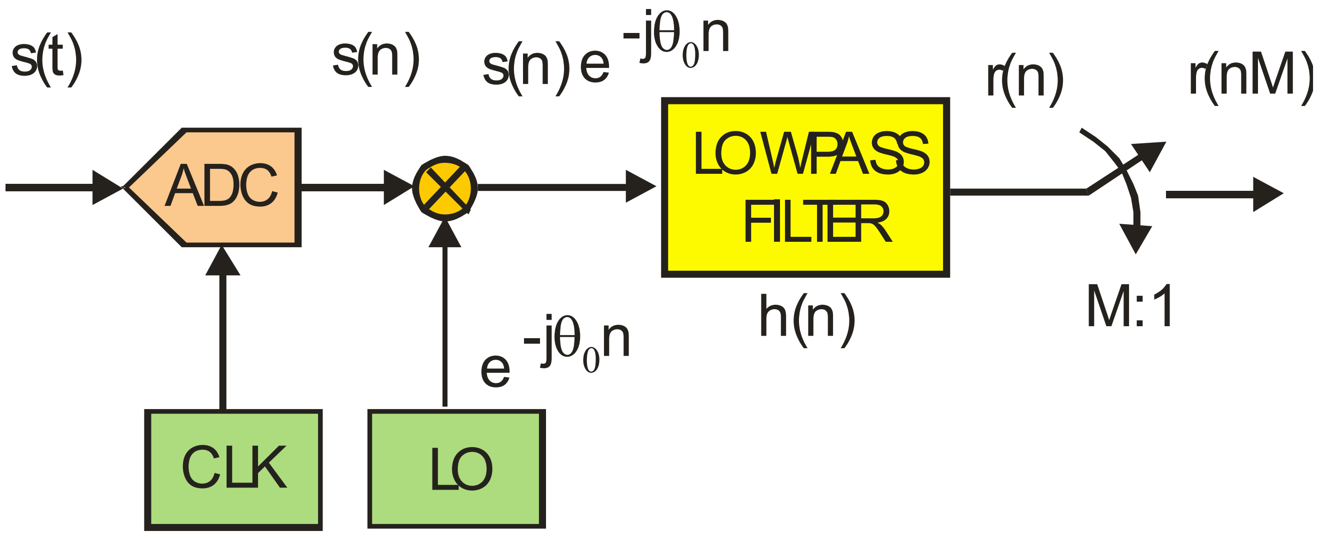


Duplicate Analog Processing in DSP

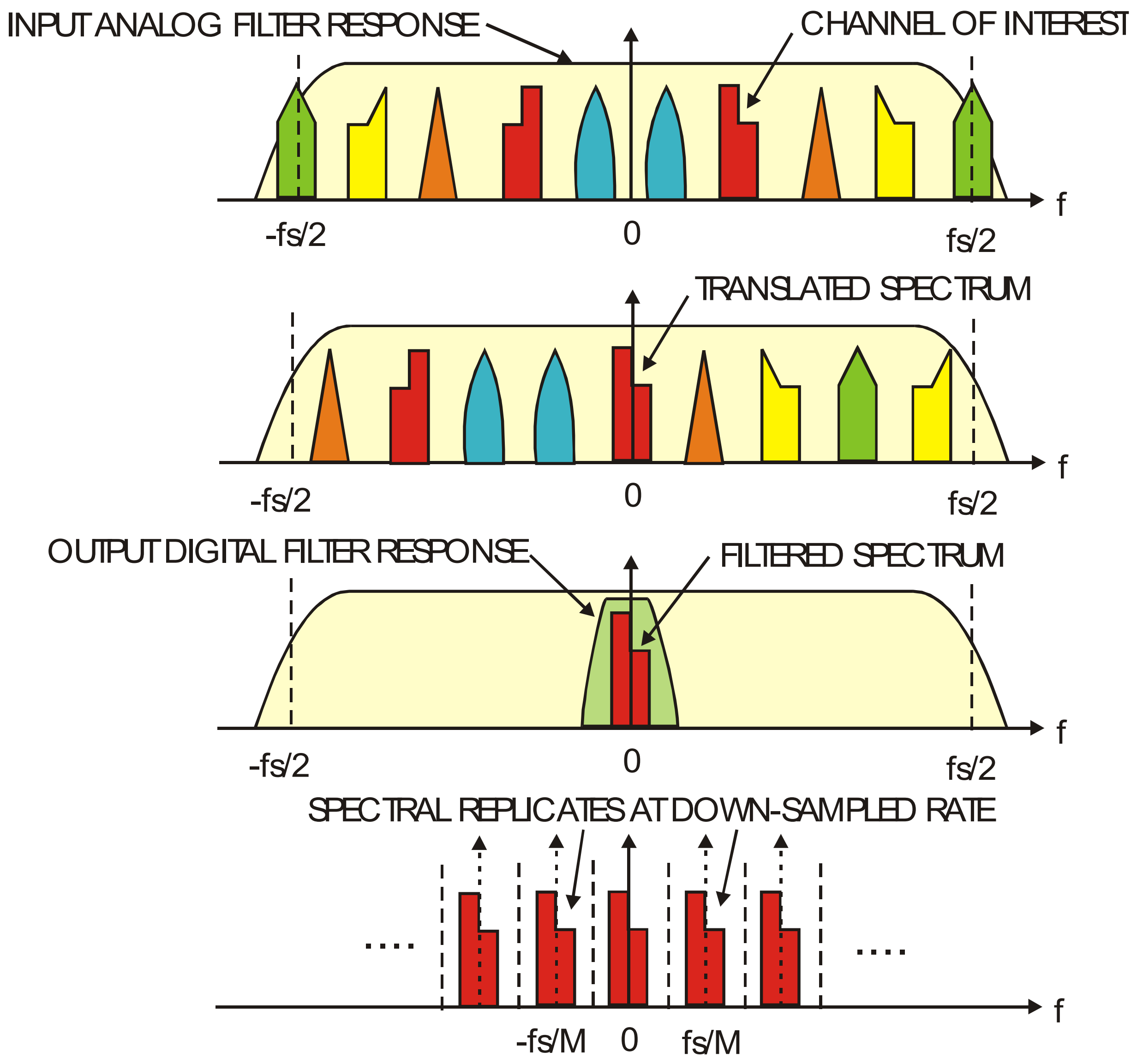


Ignoring
Good Advice!

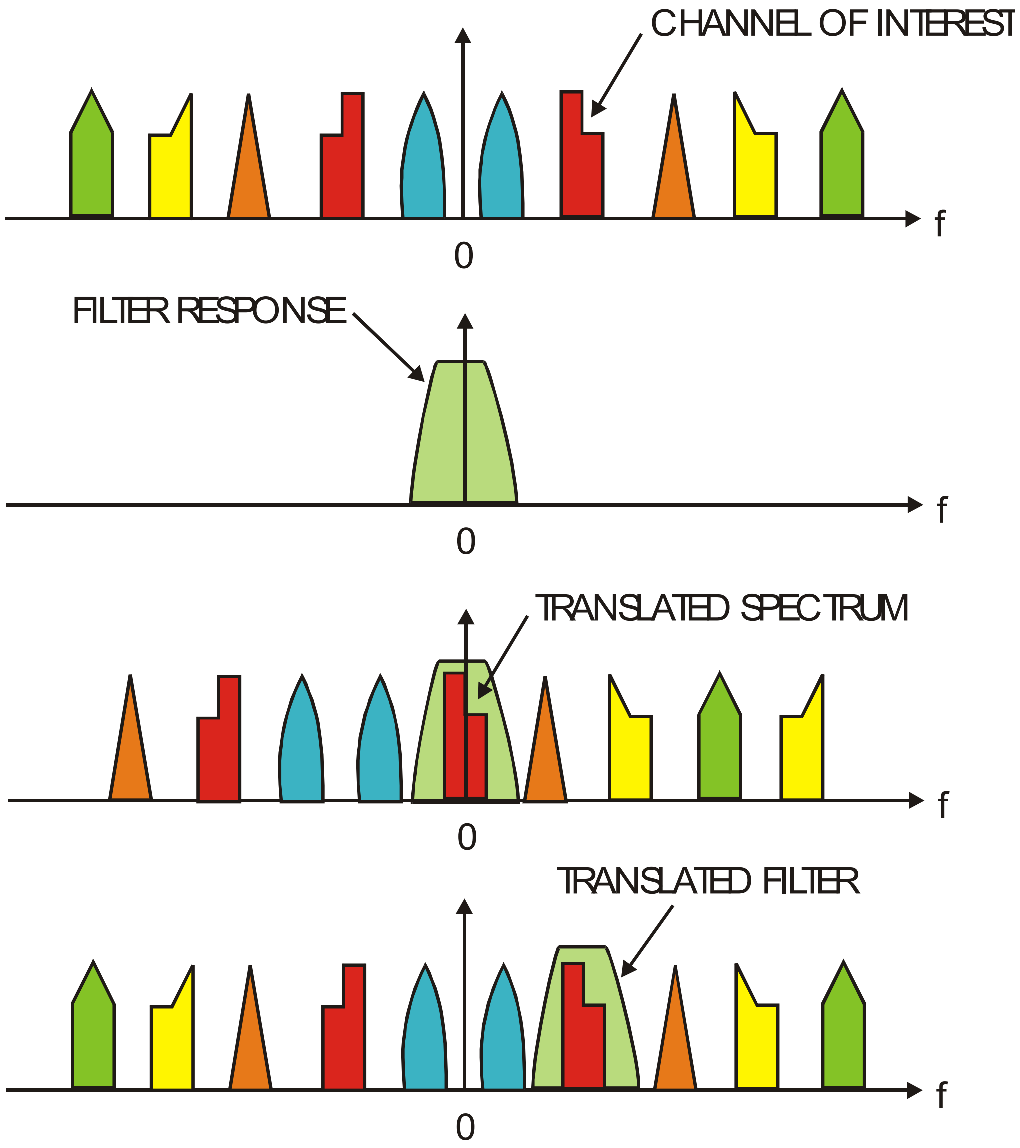
Fundamental Operations
Select Frequency, Limit Bandwidth, Select Sample Rate



Spectral Description
Fundamental Operation



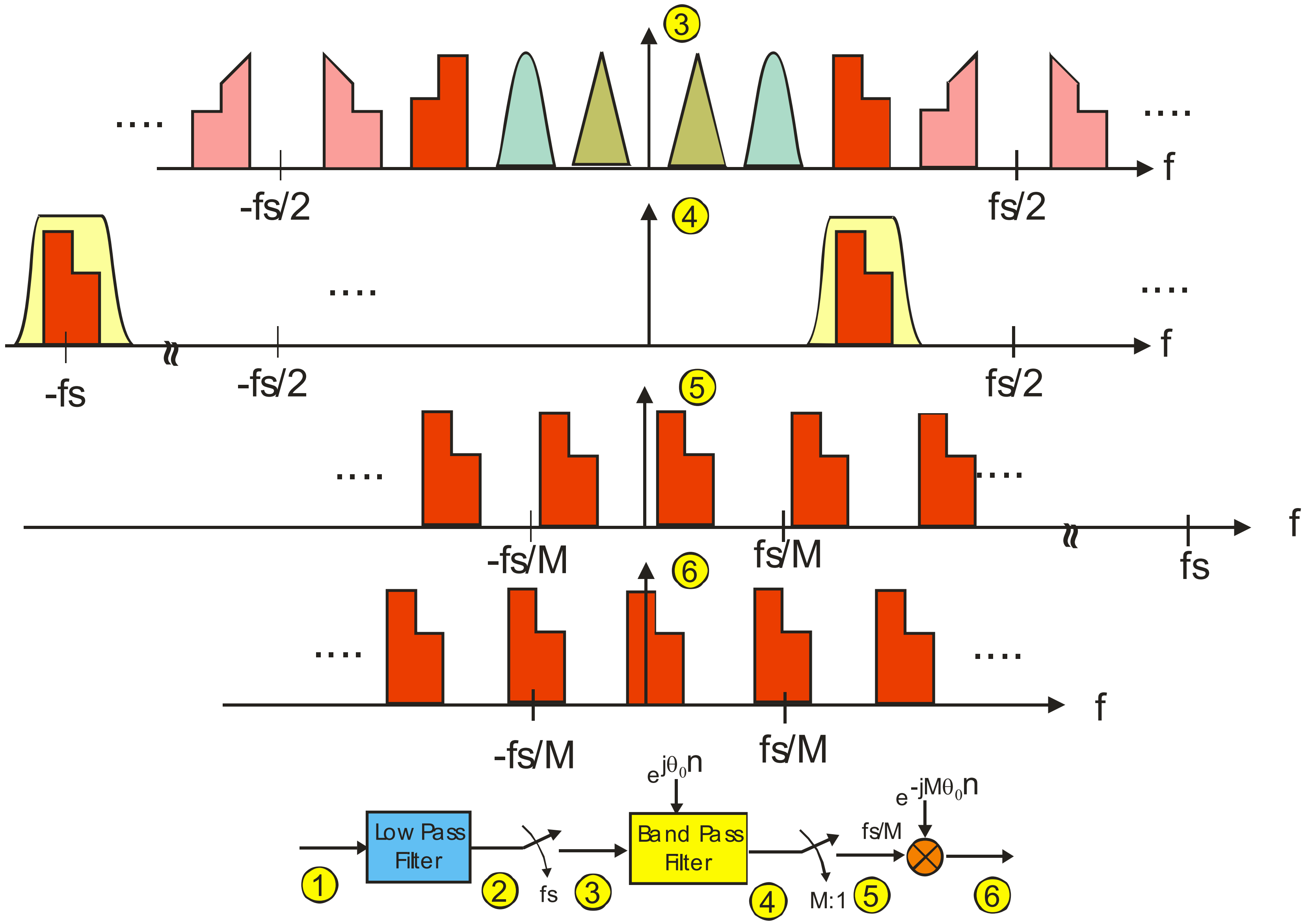
Signal and Filter are at Different Frequencies
Which One to Move??



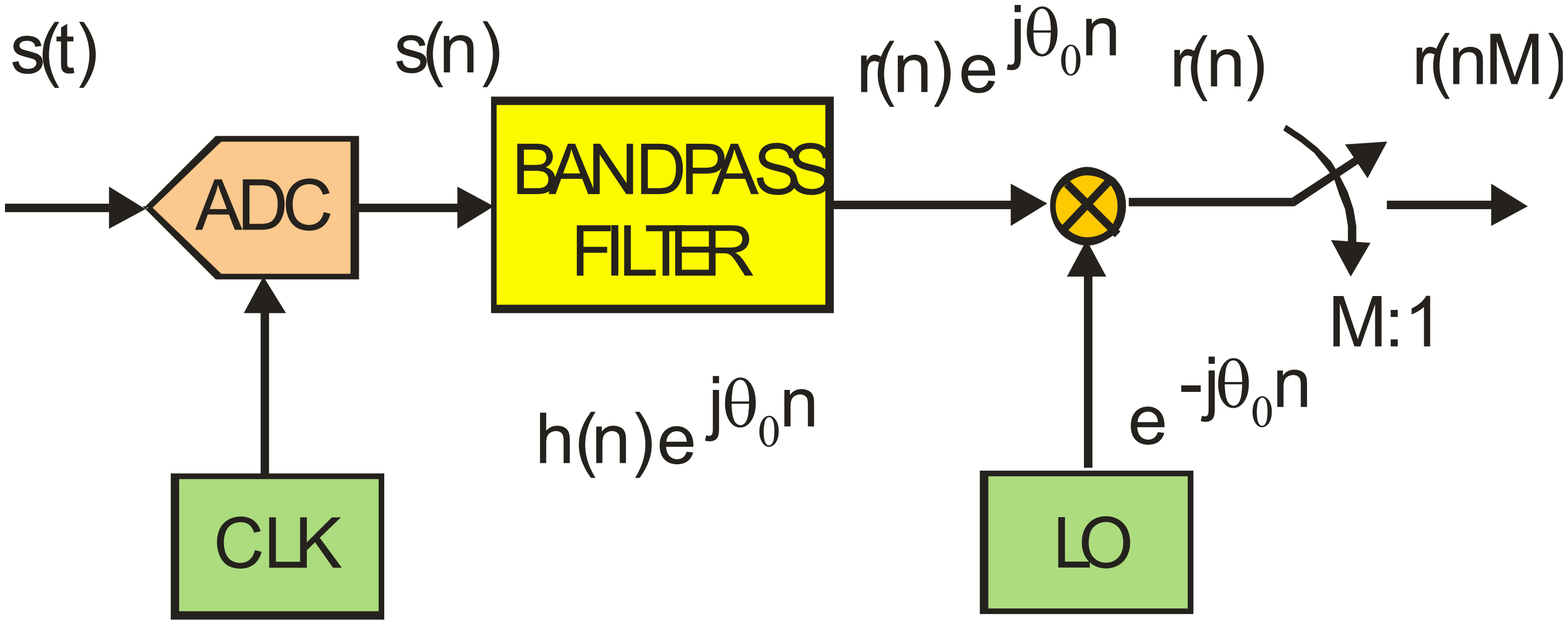
Second Option

First Option

Down Sample Complex Digital IF



Fundamental Operation Modified



Equivalency Theorem

I learned this from Irwin Jacobs in 1970
when I was a grad student at UCSD
Communication Systems
Principles of Communication Engineering
Wozencraft and Jacobs

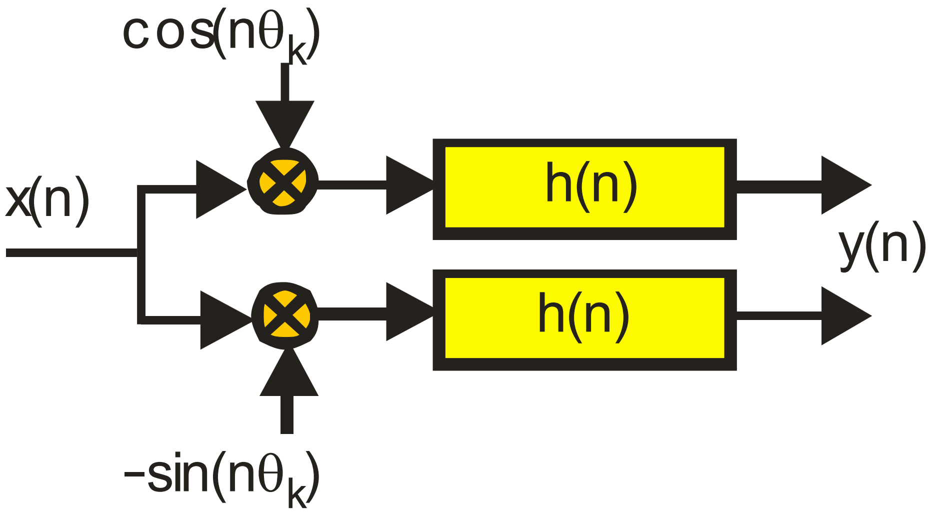
$$r(n) = s(n)e^{-j\theta_0 n} * h(k)$$

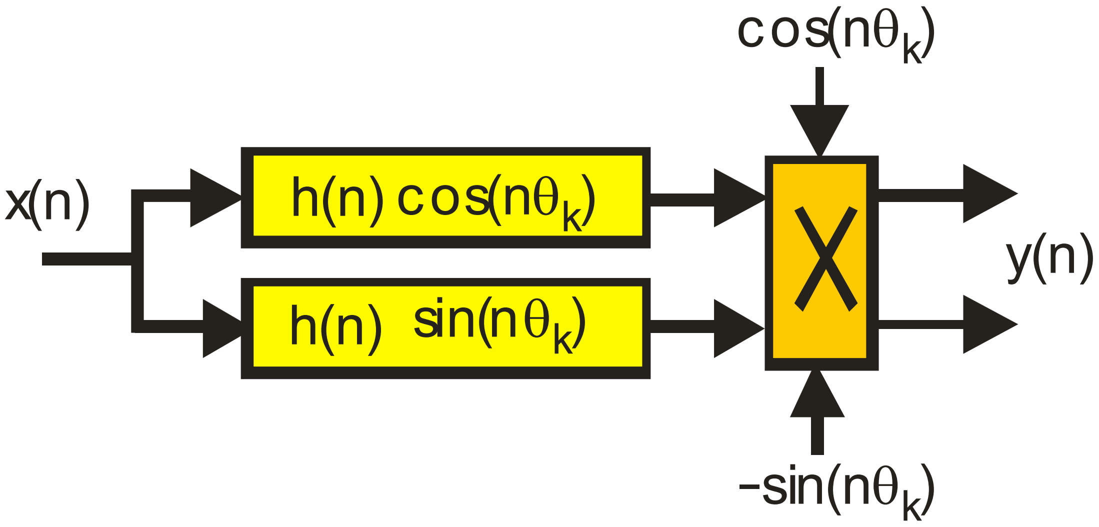
$$= \sum_k s(n-k)e^{-j\theta_0 (n-k)} h(k)$$

$$= e^{-j\theta_0 n} \sum_k s(n-k)h(k)e^{j\theta_0 k}$$

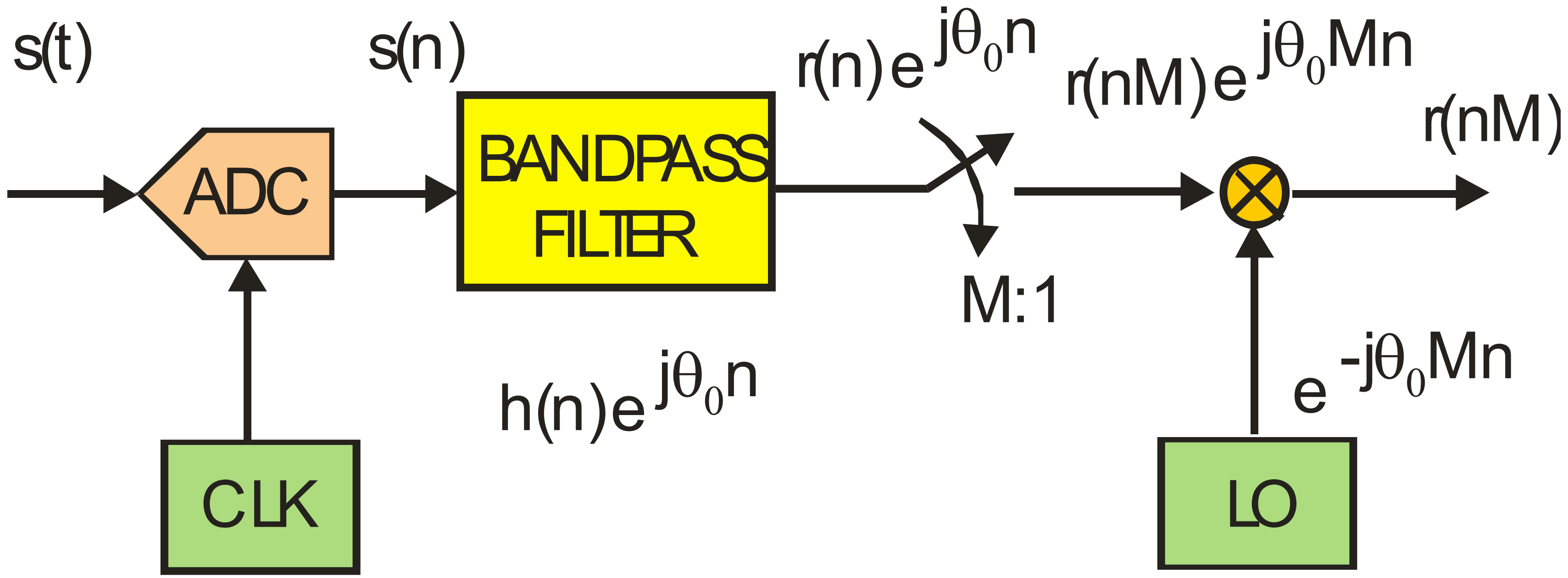
$$= e^{-j\theta_0 n} \{s(n) * h(n)e^{j\theta_0 n}\}$$

Signal Flow Description of Equivalency Theorem



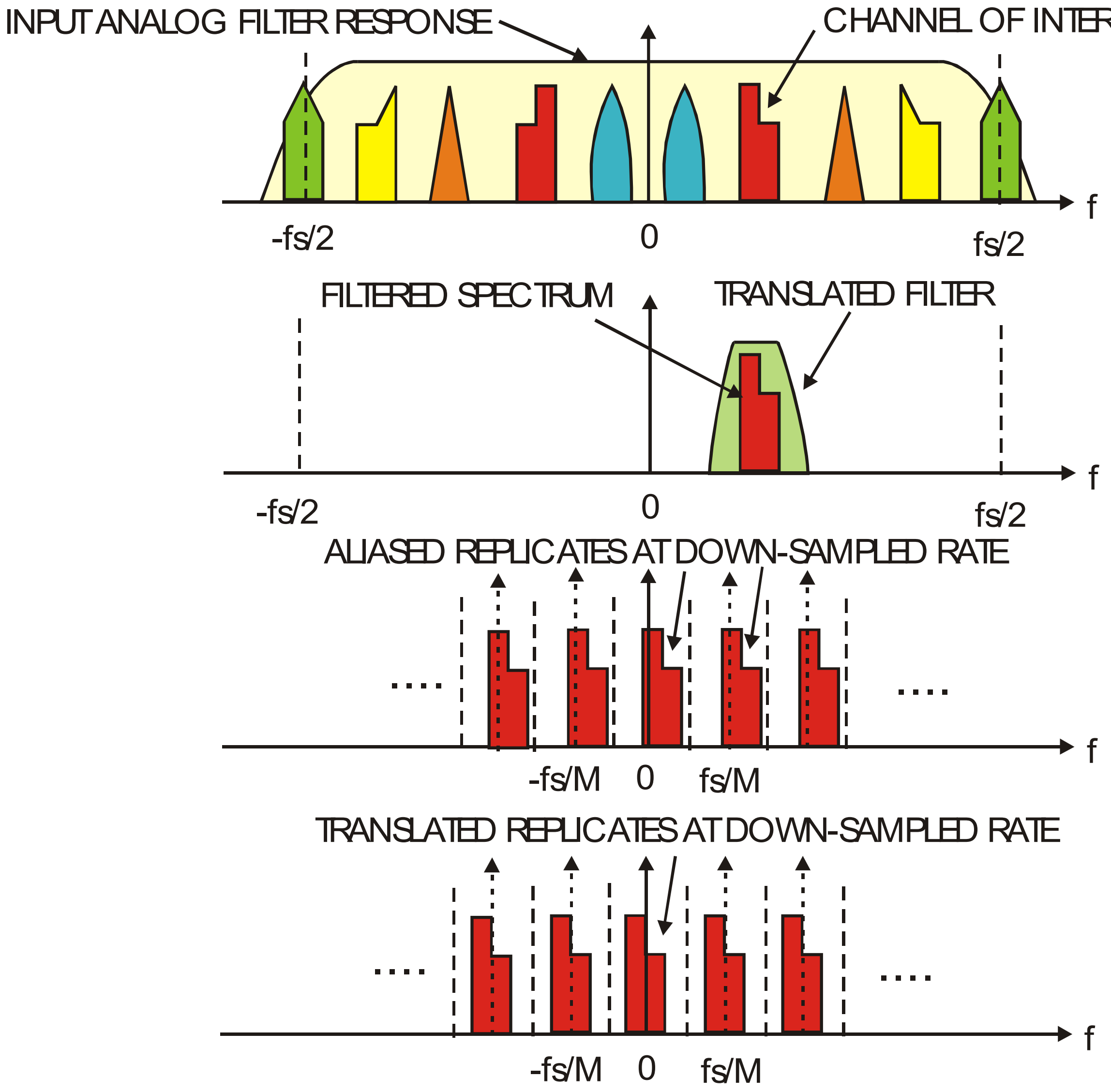


Reorder Translate and Resample

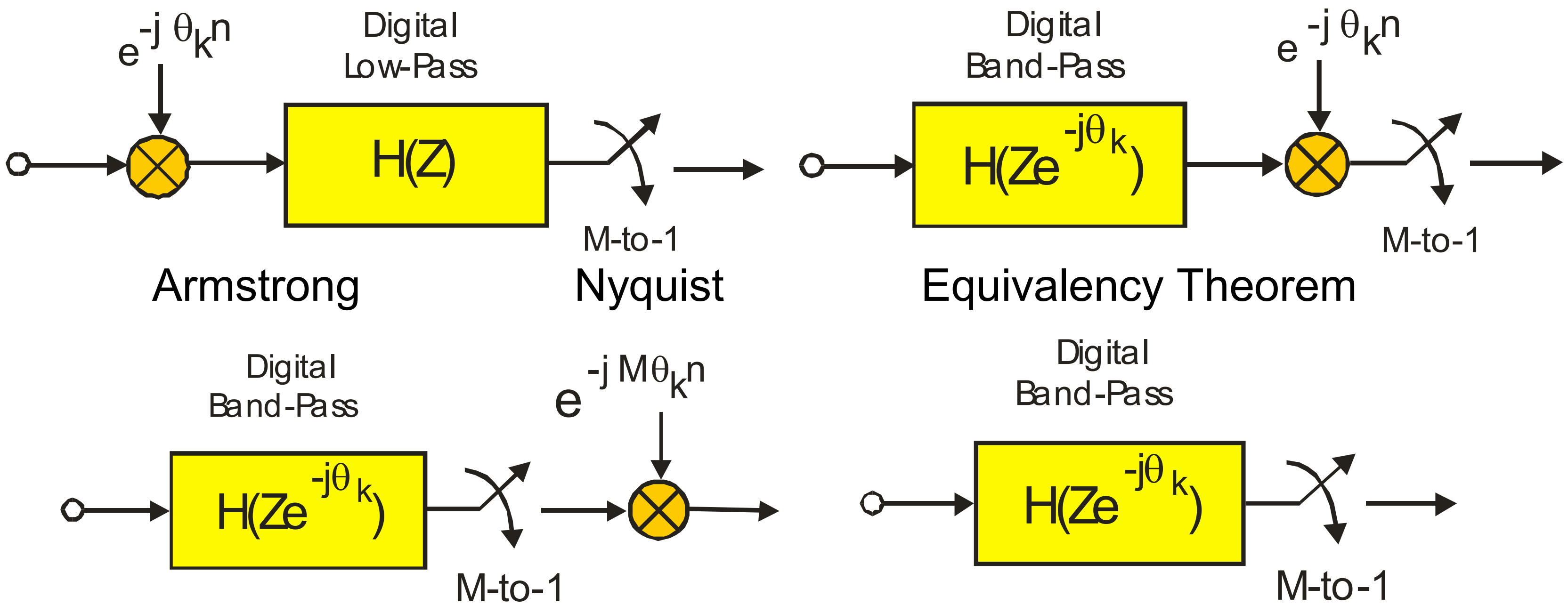


SPECTRAL DESCRIPTION

REORDERED FUNDAMENTAL OPERATION



Successive Transformations to turn Sampled Data Version of Edwin Armstrong's Heterodyne Receiver to Tuned Radio Frequency (TRF) Receiver to Aliased TRF Receiver.



Common Sense Theorem

$$M \cdot \theta_k = k \cdot 2\pi$$

$$\text{or} \quad \theta_k = k \cdot \frac{2\pi}{M}$$

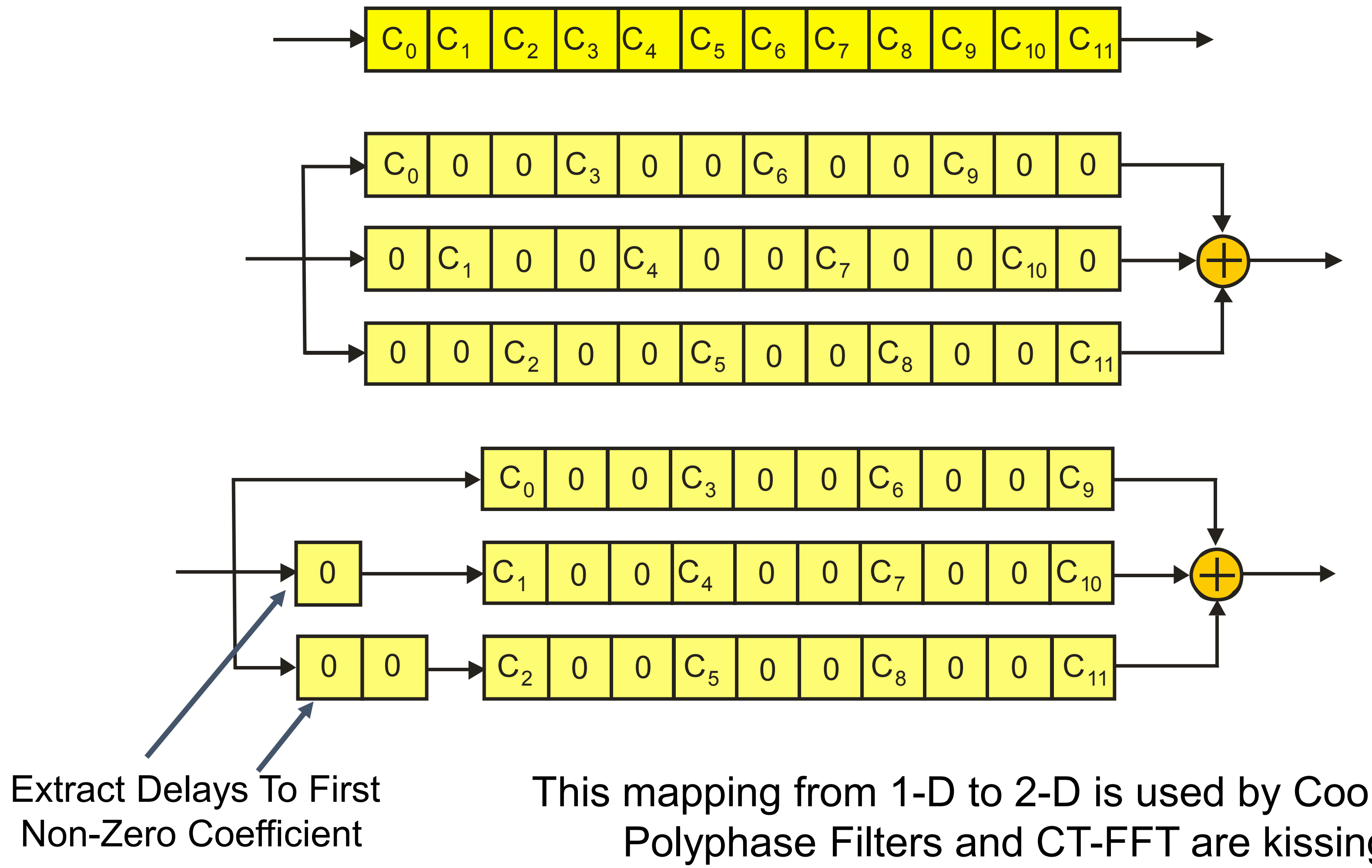
Any Multiple of
Output Sample Rate
Aliases to Baseband

Now Problem
Of computing
Output samples
Destined to be
Be discarded

Can build DSP receiver
Without a digital Down Converter

Coefficient Assignment of Polyphase Partition

For M-to-1 resample start at Index r and Increment by M
For 3-to-1 resample start at index r and increment by 3



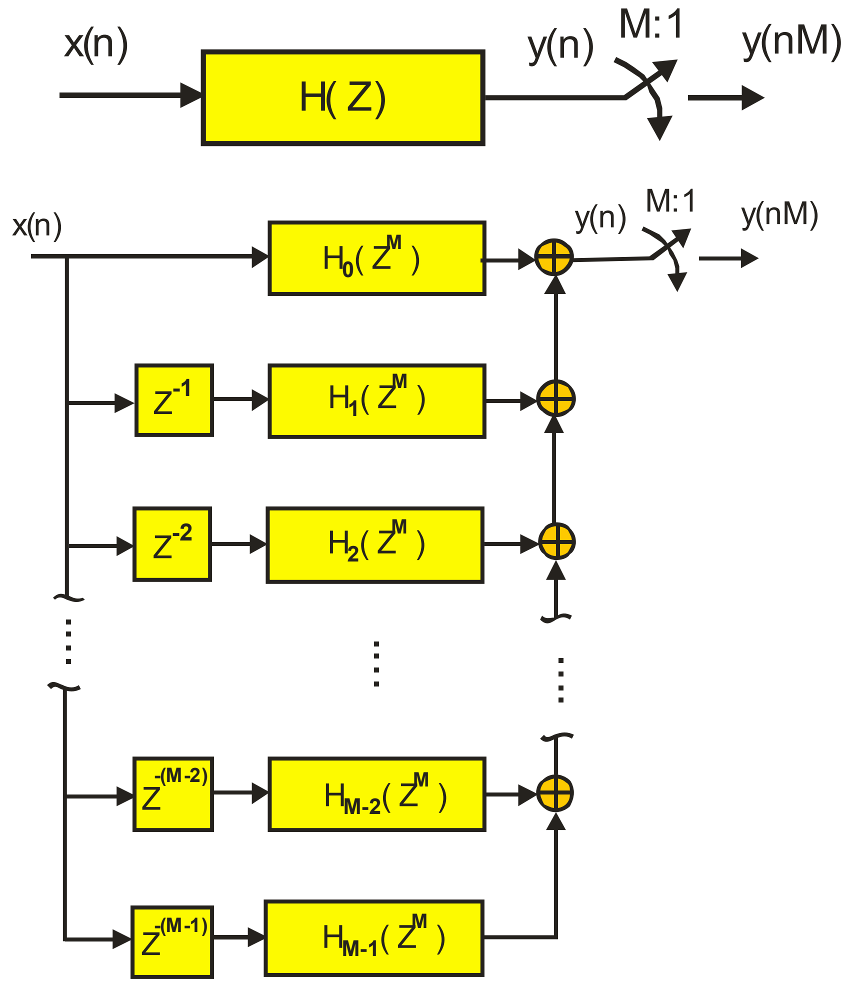
This mapping from 1-D to 2-D is used by Cooley-Tukey FFT.
Polyphase Filters and CT-FFT are kissing cousins!

Polyphase Partition of Low Pass Filter
1-Path to M-Path Transformation

$$H(Z)=\sum_{n=0}^{N-1} h(n)Z^{-n}$$

$$H(Z)=\sum_{r=0}^{M-1}\sum_{n=0}^{N-1} h(r+nM)Z^{-(r+nM)}$$

$$H(Z)=\sum_{r=0}^{M-1} Z^{-r}\sum_{n=0}^{N-1} h(r+nM)Z^{-nM}$$



M-Path Partition Supports M-to-1 Down Sample
Also Supports Rational Ratio
M-to-Q and M-to-Q/P Down Sample!

Polyphase Partition of Band Pass Filter 1-Path to M-Path Transformation

Modulation Theorem of Z-Transform

$$G(Z) = \sum_{n=0}^{N-1} h(n)\, e^{j\theta_k n} Z^{-n} = \sum_{n=0}^{N-1} h(n)\,(e^{-j\theta_k} Z)^{-n} = H(e^{-j\theta_k} Z)$$

$$G(Z) = \sum_{r=0}^{M-1} \sum_{n=0}^{N-1} h(r+nM)\, e^{j\theta_k (r+nM)} Z^{-(r+nM)}$$

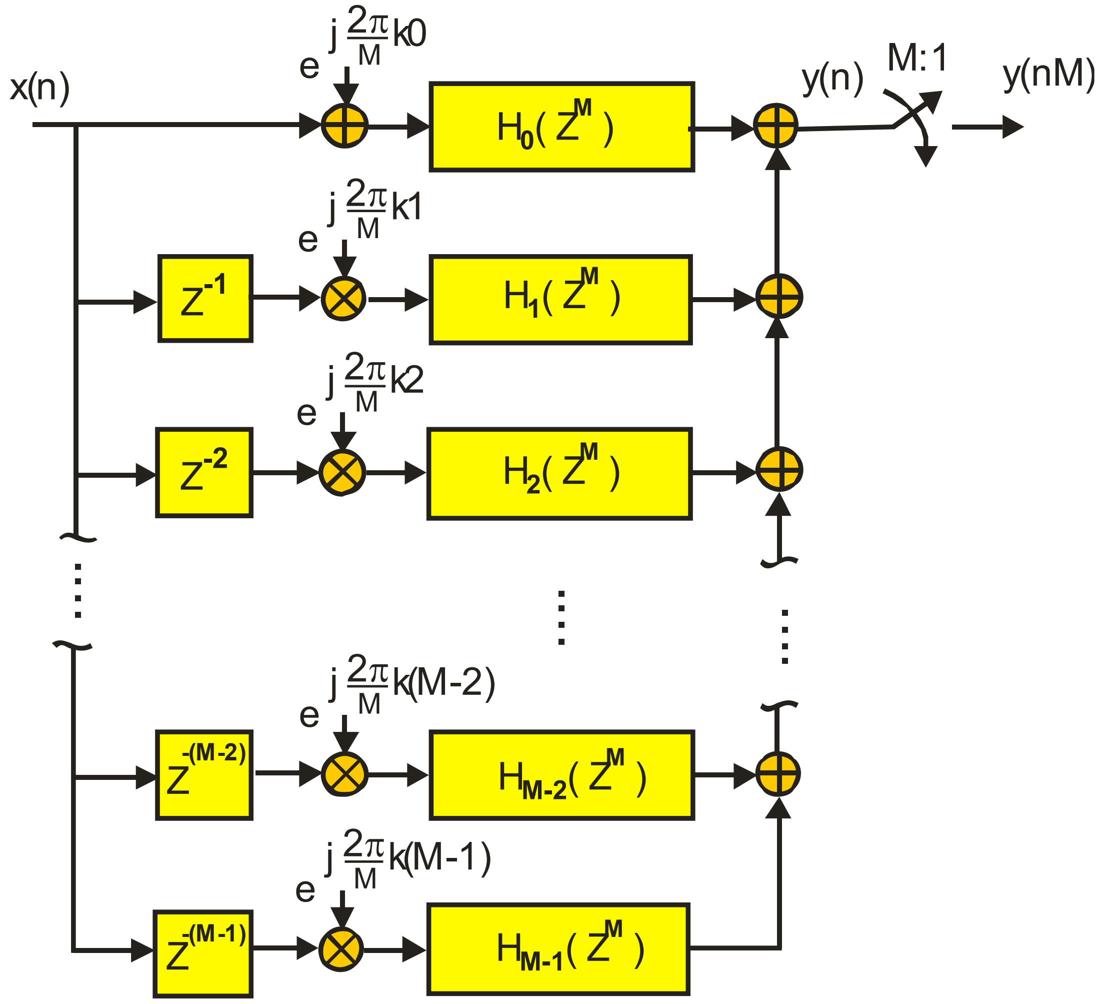
$$G(Z) = \sum_{r=0}^{M-1} e^{j\theta_k r} Z^{-r} \sum_{n=0}^{N-1} h(r+nM) e^{j\theta_k nM} Z^{-nM}$$

$$G(Z) = \sum_{r=0}^{M-1} e^{j\frac{2\pi}{M}kr} Z^{-r} \sum_{n=0}^{N-1} h(r+nM) Z^{-nM}$$

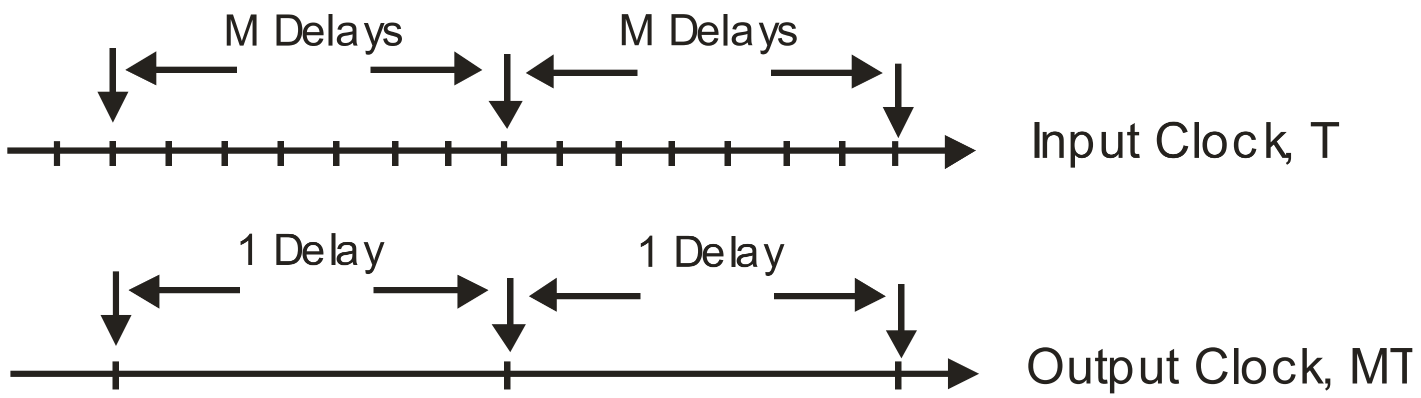
$$\mathrm{M} \cdot \theta_k = k \cdot 2\pi$$

$$\text{or} \quad \theta_k = k \cdot \frac{2\pi}{\mathrm{M}}$$

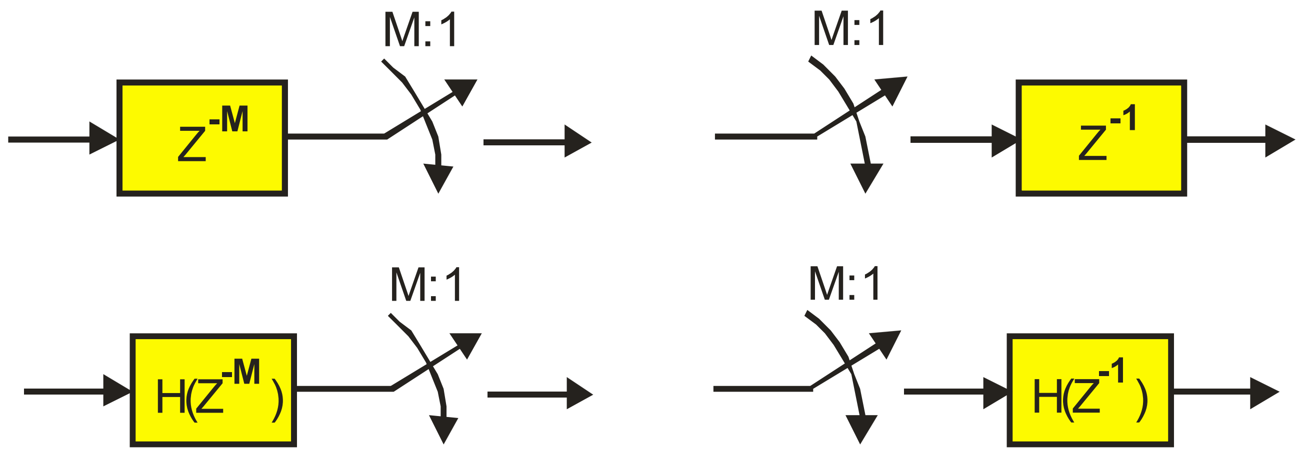
Polyphase Band Pass Filter and M-to-1 Resampler



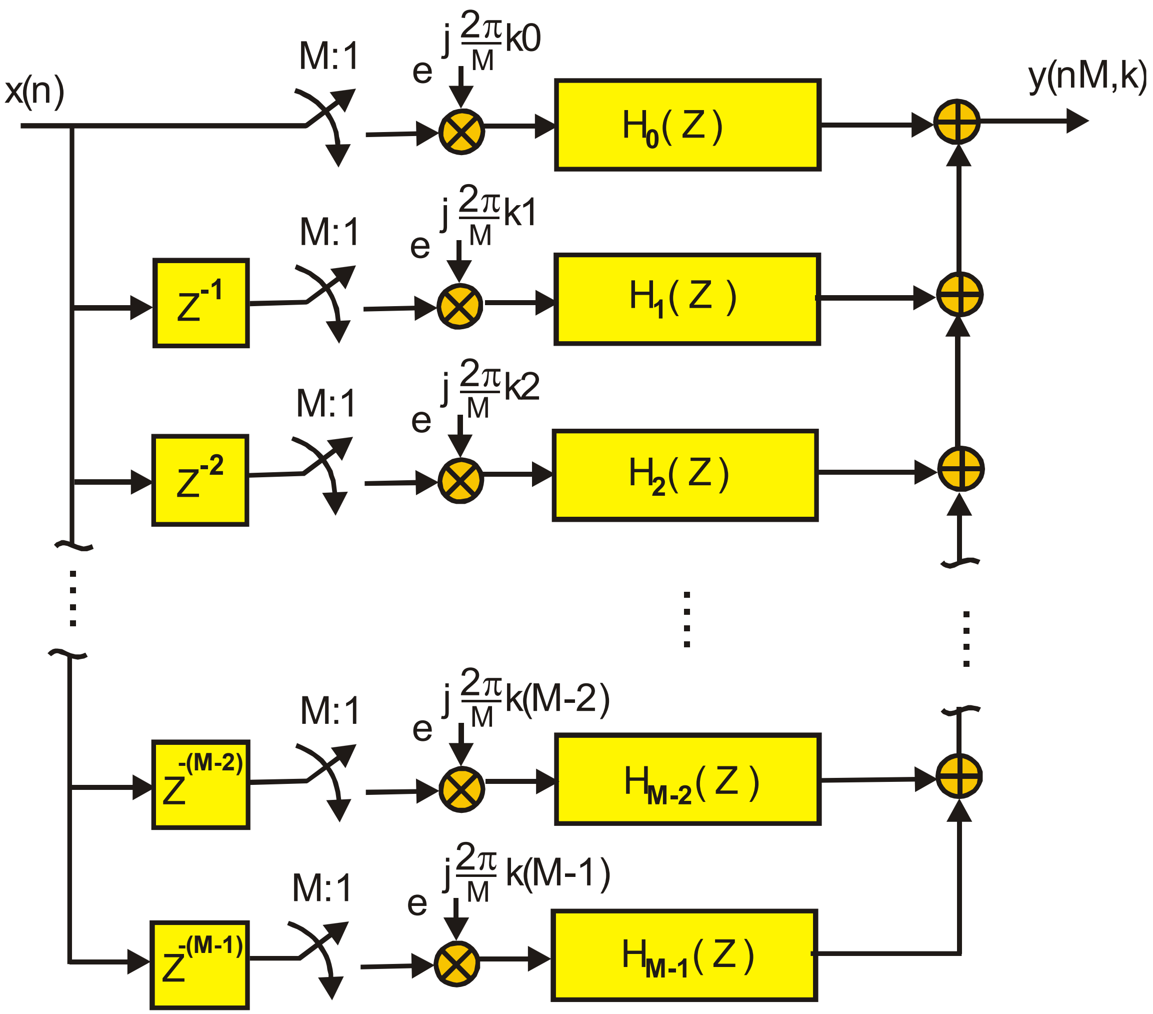
Noble Identity: Commute M-units of Delay followed by M-to-1 Down Sample



M-Units of Delay at Input Rate Same as 1-Unit of Delay at Output Rate



Apply Noble Identity to Polyphase Partition



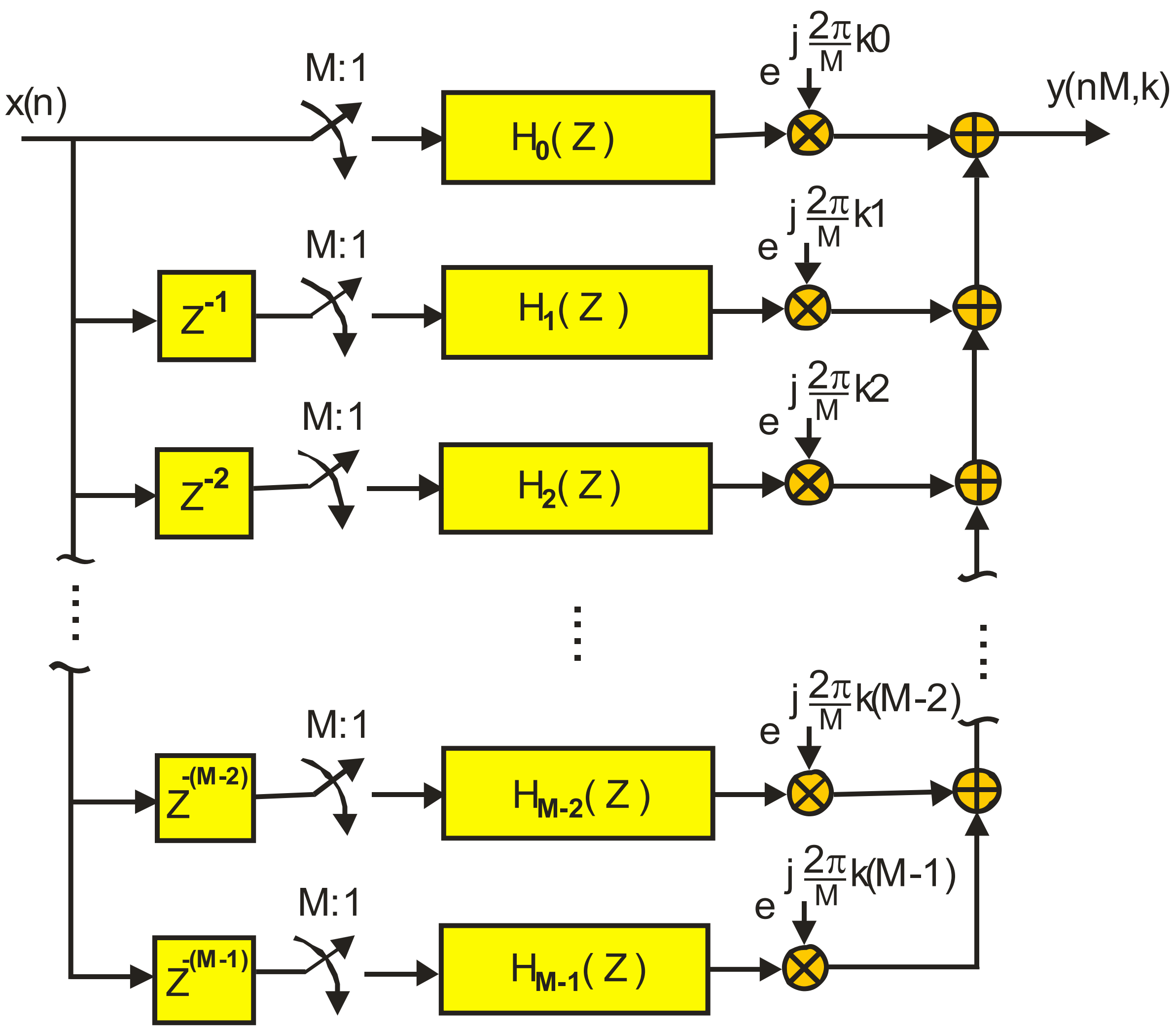
We Reduce Sample Rate
M-to-1 Prior to Reducing Bandwidth

(Nyquist is Raising His Eyebrows!)

We Intentionally Alias the Spectrum.
(Were you Paying Attention
in school when they discussed the
importance of anti-aliasing filters?)

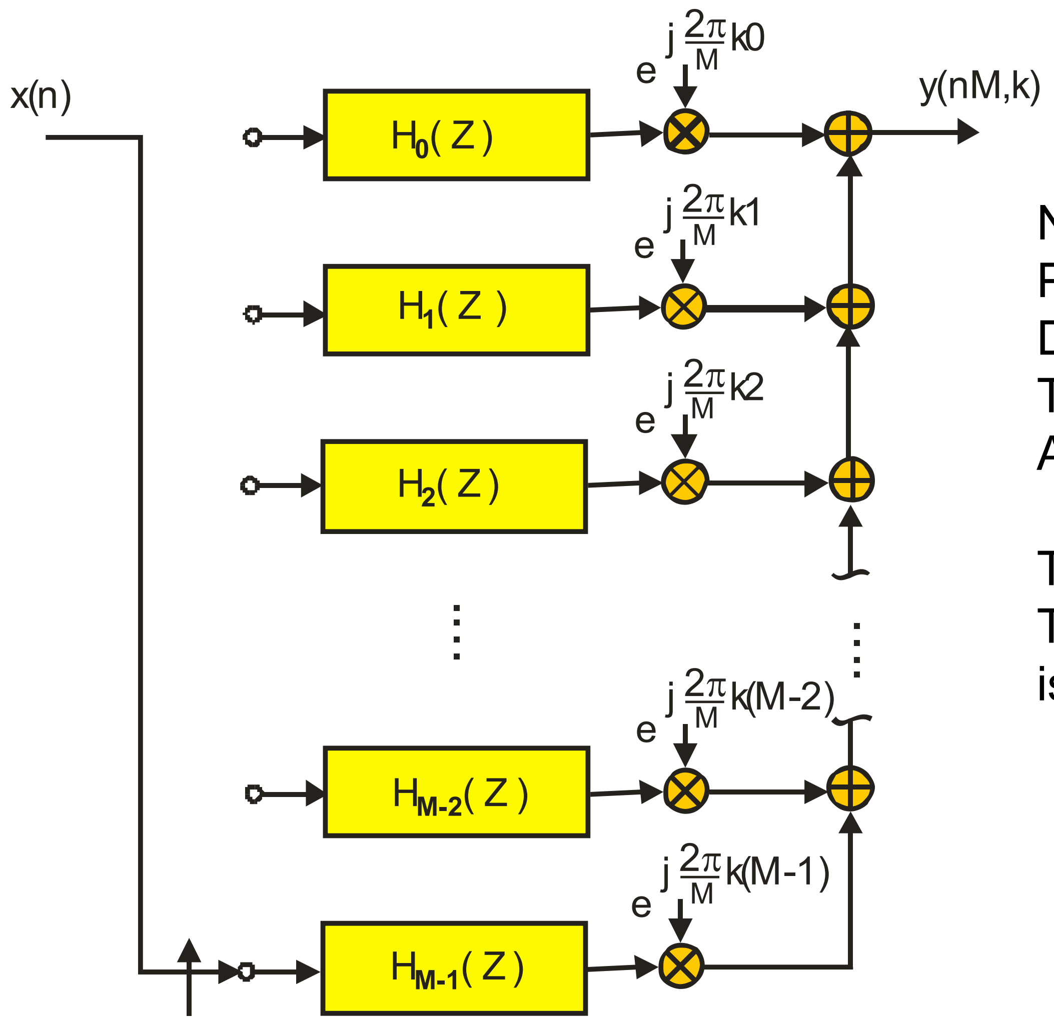
M-fold Aliasing!
M-Unknowns!
M-Paths supply M-Equations
We can the separate Aliases!

Move Phase Spinners to Output of Polyphase Filter Paths



Want Phase Spinners as far away from resampler as possible

Polyphase Partition with Commutator Replacing the “r” Delays in the “r-th” Path

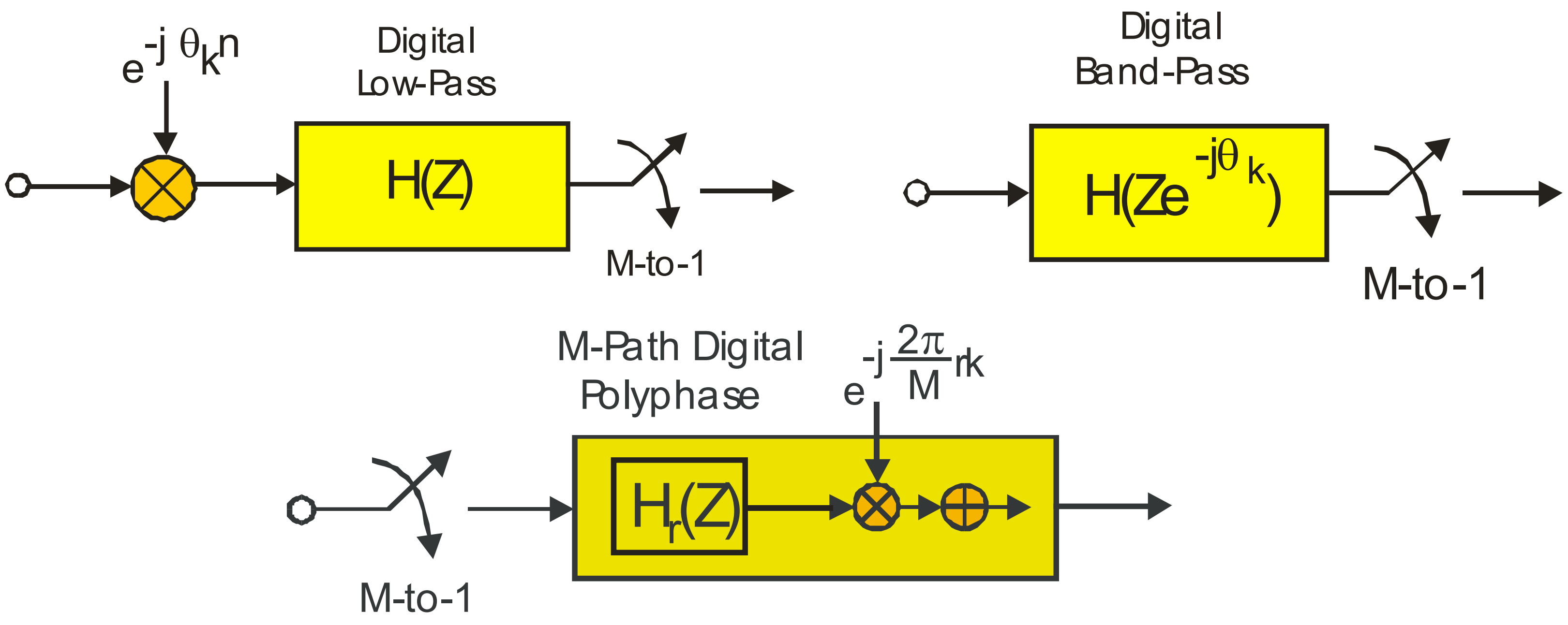


Note: We don’t assign Phase Spinners to Select Desired Center Frequency Till after Down Sampling And Path Processing

This Means that The Processing for every Channel is the same till the Phase Spinners

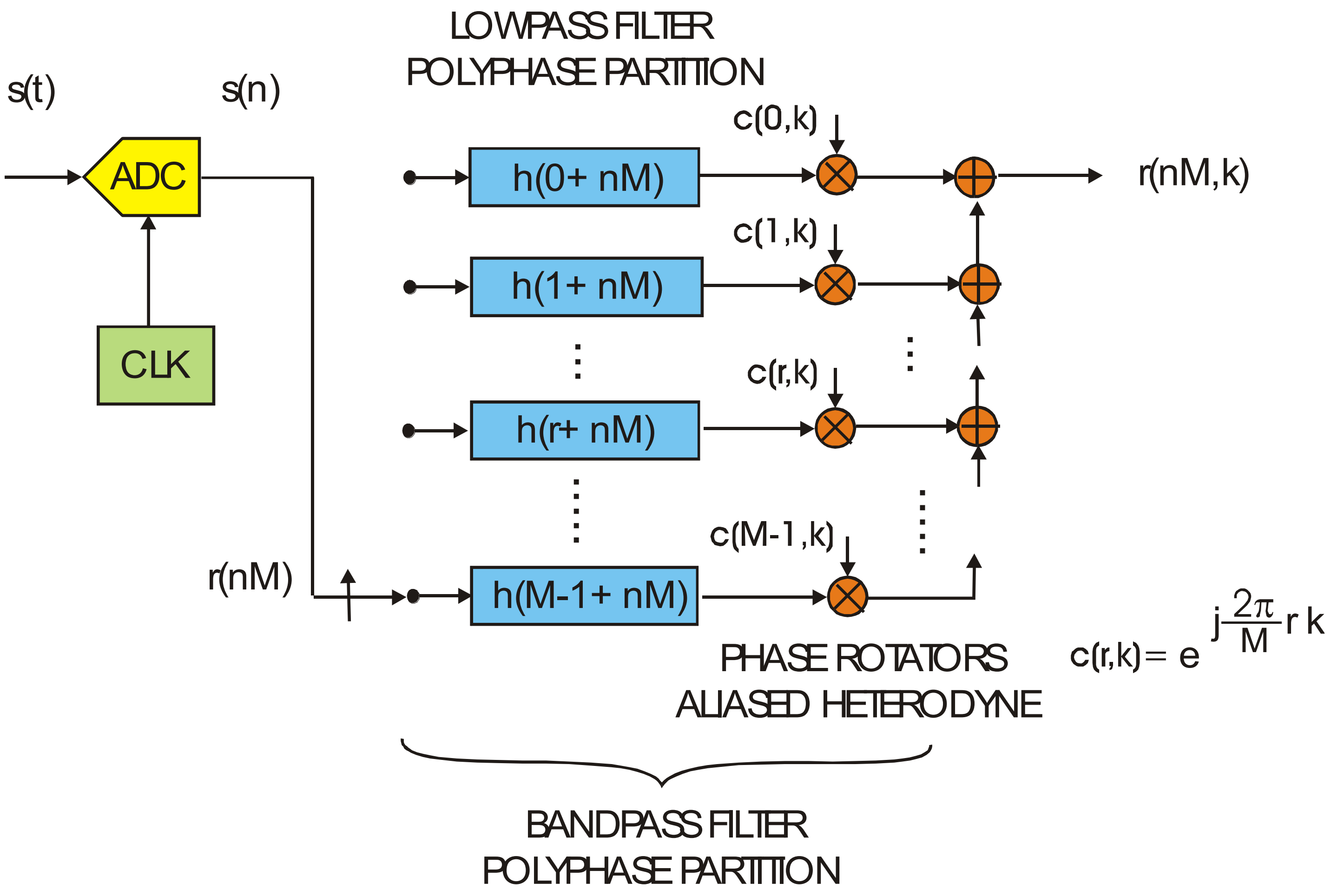
No longer LTI, Filter now has M-Different Impulse Responses! Now LTV or PTV Filter.

Armstrong to Tuned RF with Alias Down Conversion to Polyphase Receiver



Rather than selecting center frequency at input and reduce sample rate at output, we reverse the order, reduce sample rate at input and select center frequency at output. We perform arithmetic operations at low output rate rather than at high input rate!

Reorder Filter and Resample



A Sad but True Story!

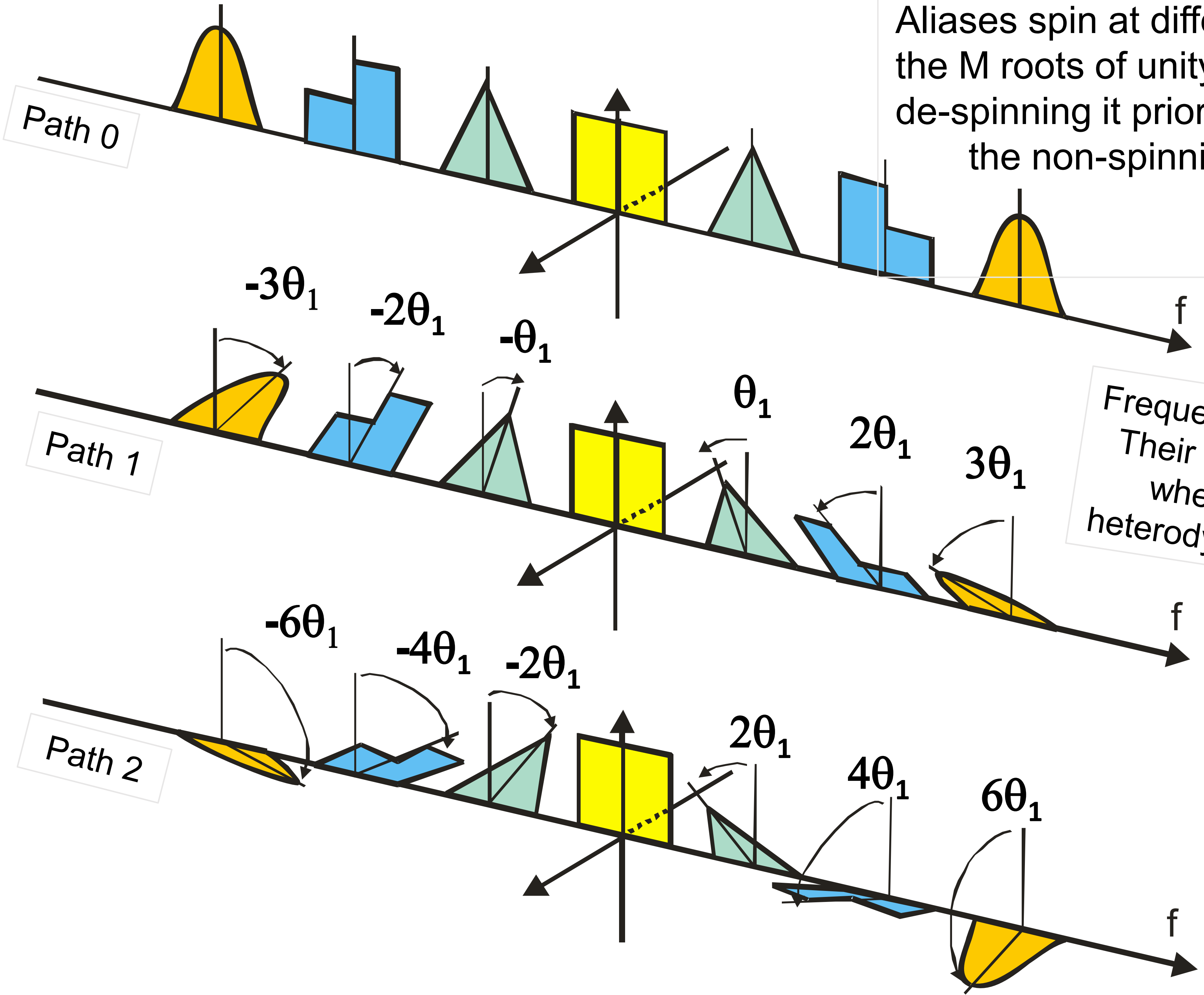
- In 1983 I designed a 65536 channelizer for GTE in San Jose in response to a Request for Proposal (RFP) for Multi channel receiver for US Navy.
- My Design aliased all 65 k channels to baseband by down sampling prior to any signal processing. I then used the phase rotators of the IFFT to extract (un-alias) the separate channels.
- I was sure we would be famous. We were revolutionizing channelizers! And GTE was assured to be awarded the contract!
- The proposal was returned to us by the reviewer

  with **big red** letters across the face of the proposal saying…

-  "Those Who Don't Understand The Nyquist Theorem Shouldn't be Doing Signal Processing"

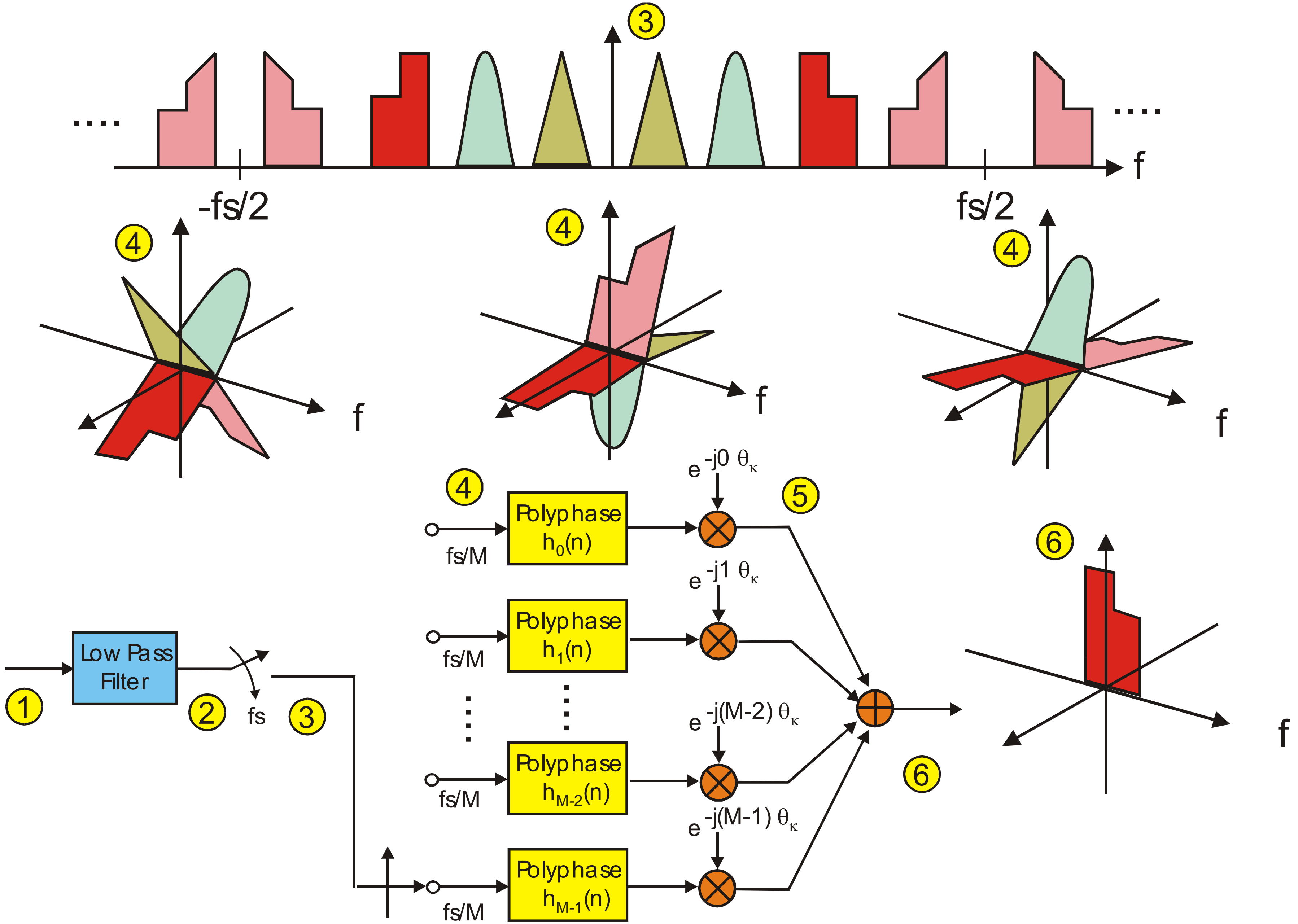
What I Learned is that if you are smarter than your reviewer, you are both in trouble.

Aliases spin at different rates. All multiples of the M roots of unity. Extract a particular alias by de-spinning it prior to summing the paths. Only the non-spinning alias survives summation. All other aliases destructively cancel!

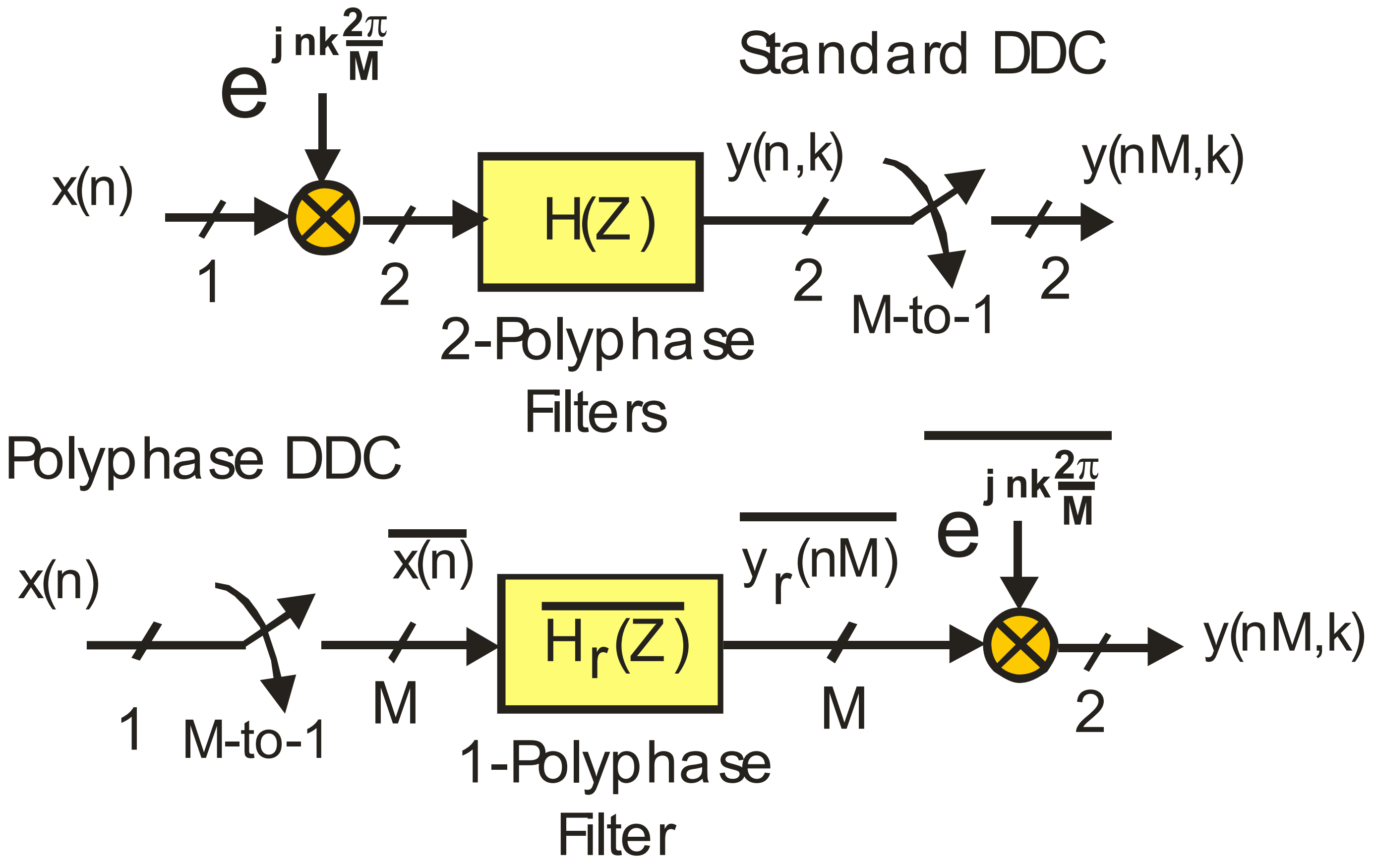


Frequency bands preserve Their complex envelope when translated by heterodyne or by aliasing!

Polyphase Partition

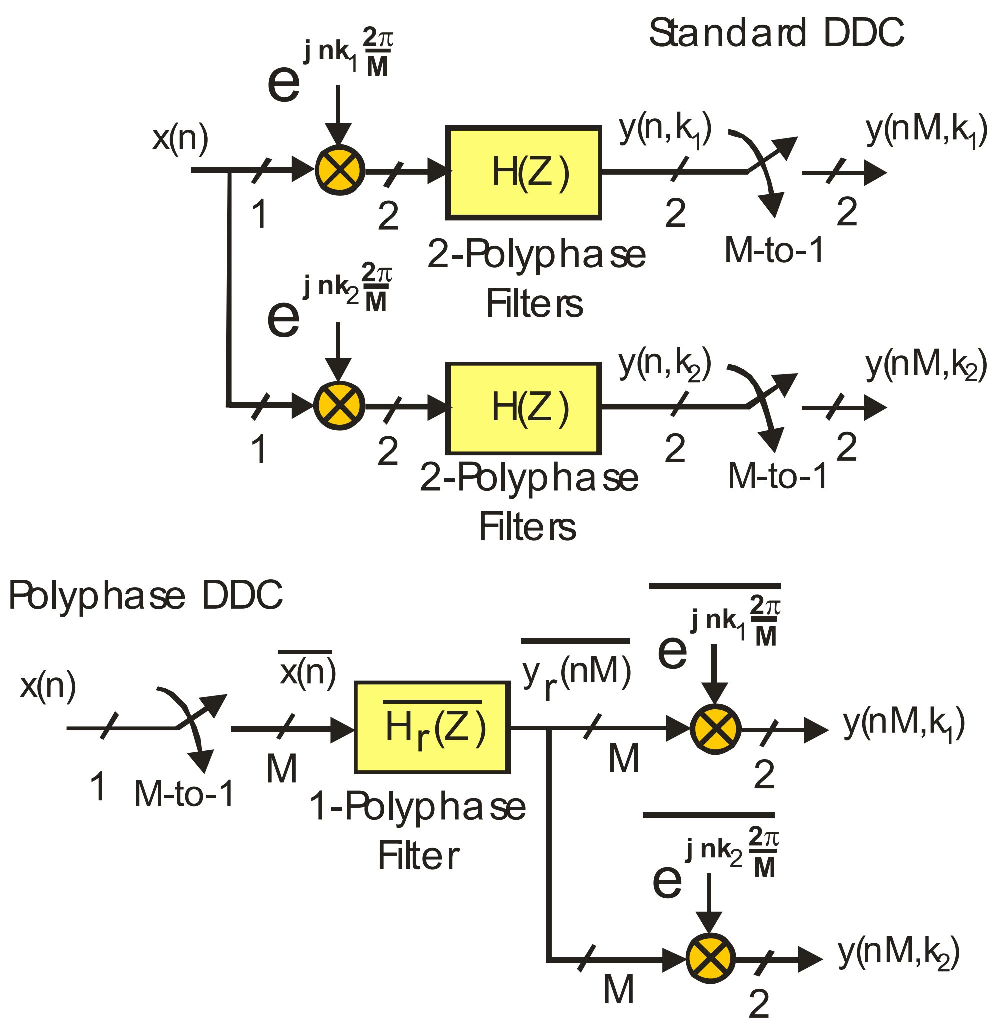


Single Channel Armstrong and Multirate Aliased Polyphase Receiver

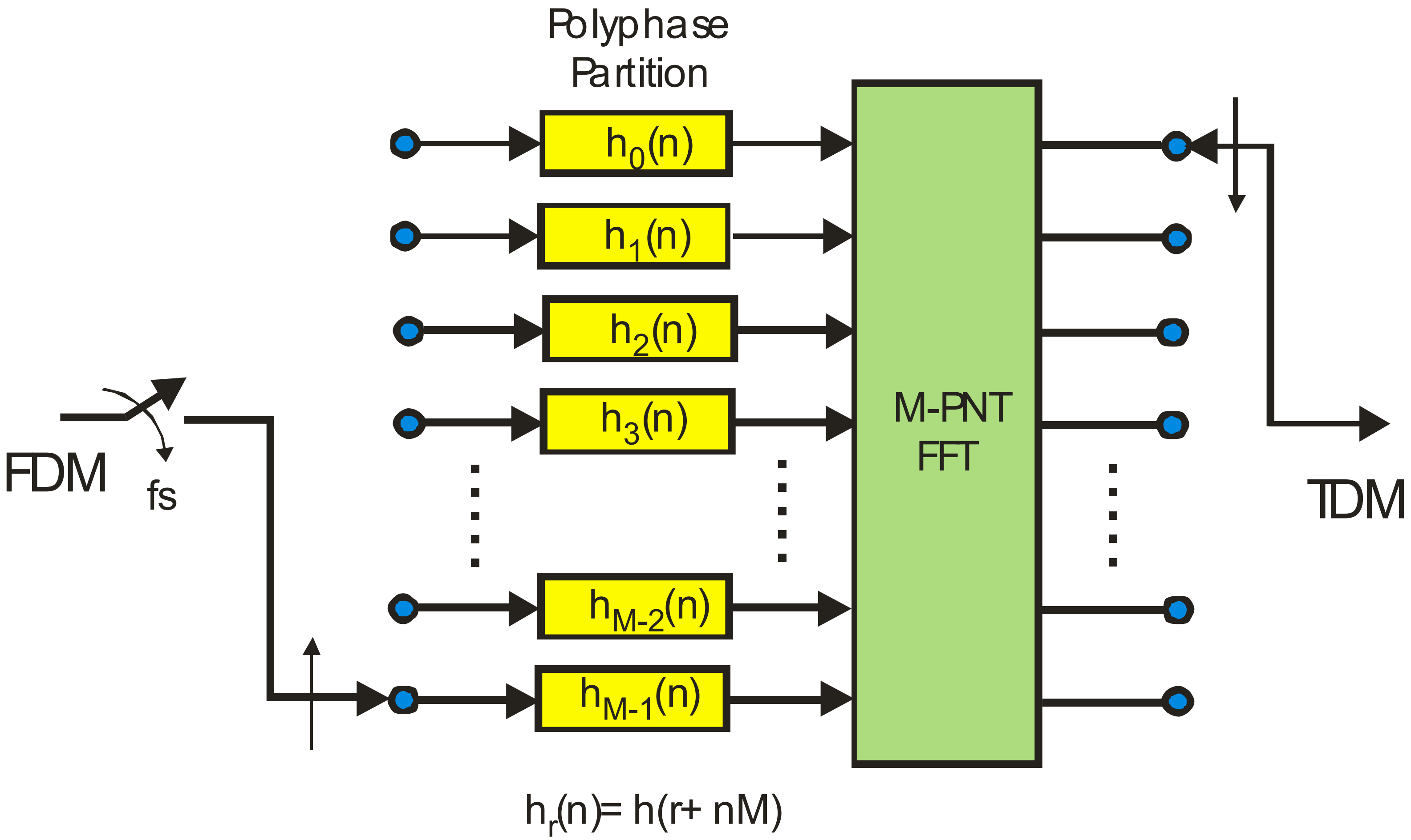


Rather than select the band you want to move to baseband (Armstrong heterodyne), you move all bands to baseband and extract the alias of interest by phase coherent Summation which destructively cancels all other aliases.

Dual Channel Armstrong and Multirate Aliased Polyphase Receiver



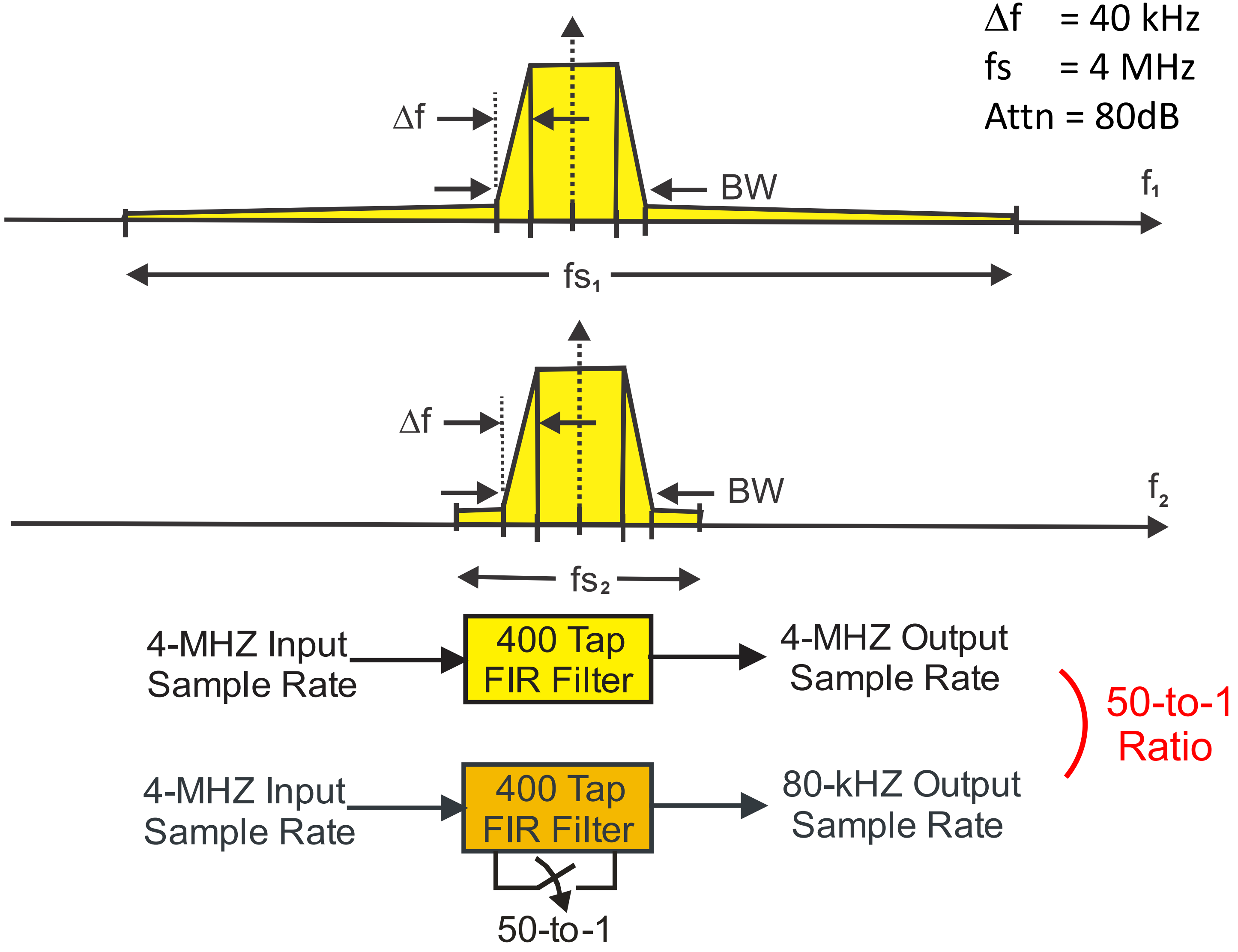
M-Channel Polyphase Channelizer: M-path Filter and M-point FFT



Advantage of Polyphase Filter over single stage tapped delay line implementation

Sample Rate Large Compared to Bandwidth

BW = 40 kHz
Δf = 40 kHz
fs = 4 MHz
Attn = 80dB



$$N_1 \cong \frac{f_s}{\Delta f}\,\frac{Attn(dB)}{20}$$

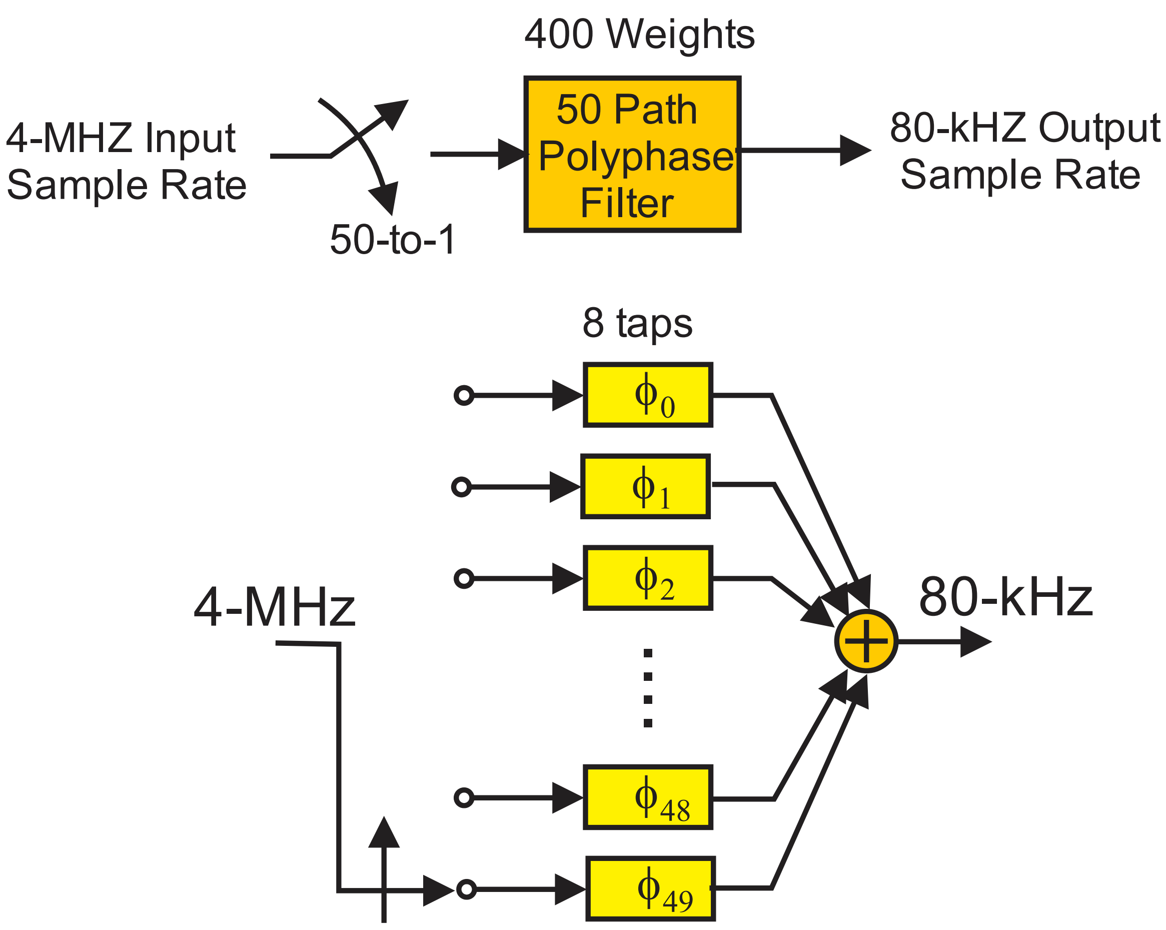
$$N_1 = \frac{4000}{40}\,\frac{80}{20} = 400 \text{ Taps}$$

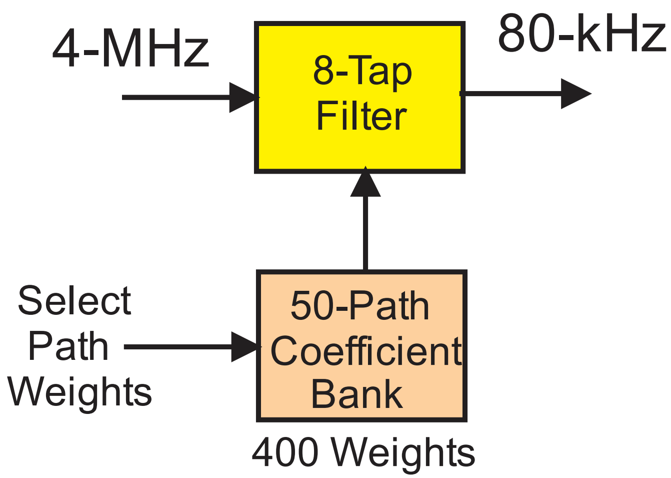
$$\min f_s = 80 \text{ kHz}$$

$f_S > BW + \Delta f$
(2 sided BW + transition BW

Nyquist Rate for Filter is 80 kHz or fs/50
Can Perform 50-to-1 Down Sample and Still Satisfy Nyquist

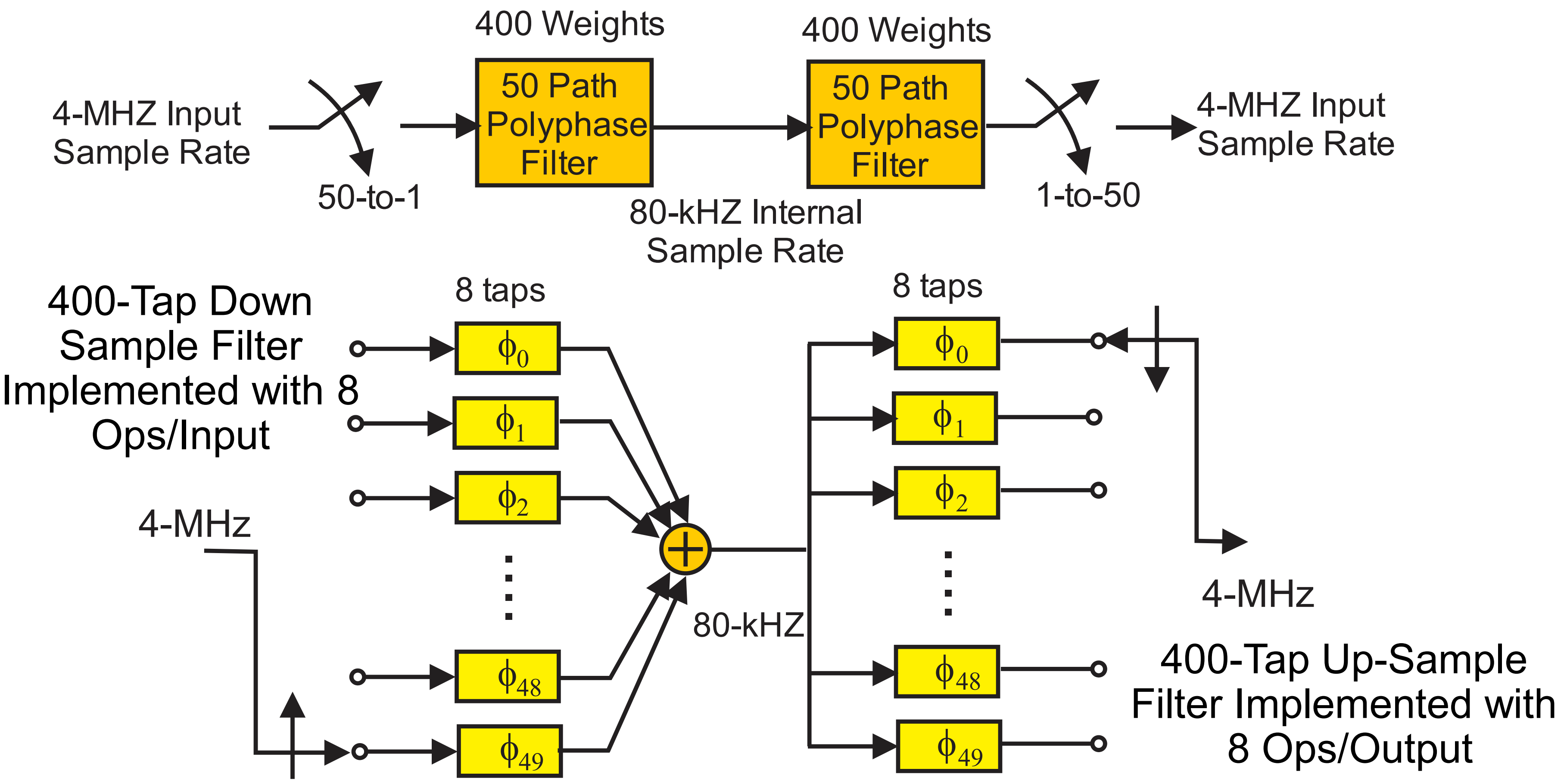
Reduce Sample Rate at Input to Filter: Very Efficient Implementation!





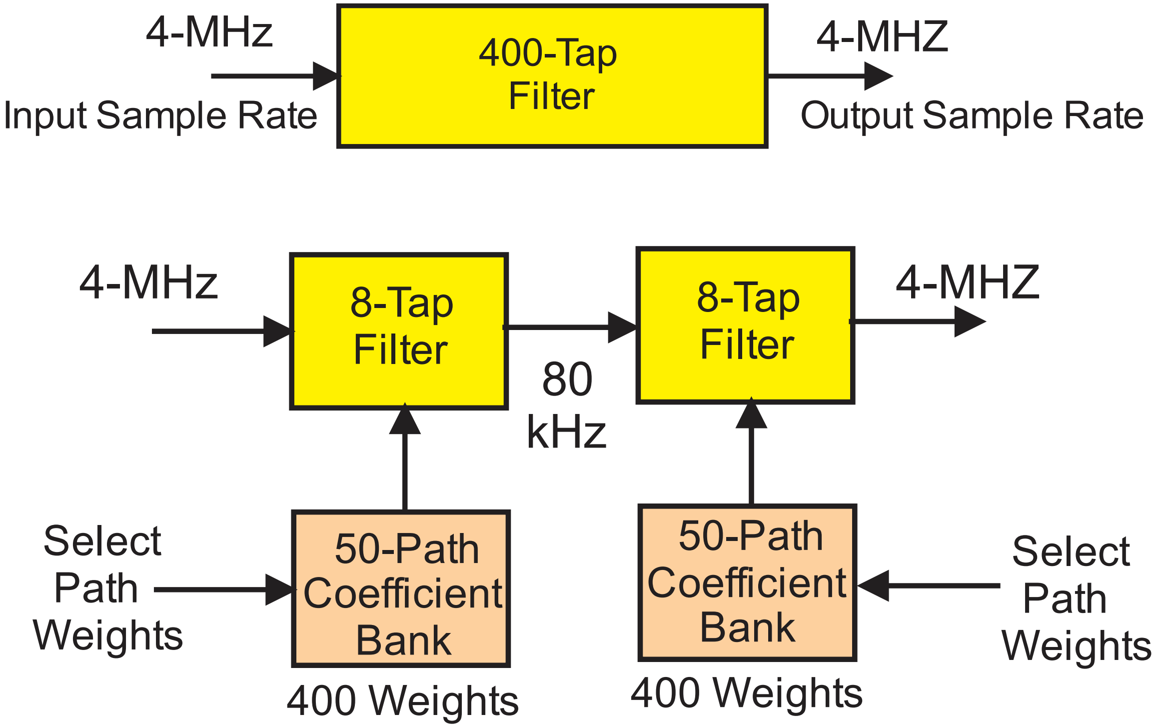
N/M = 400/50
= 8-ops/Input

Down Sample to Reduce Sample Rate in Proportion to Bandwidth Reduction and Up Sample to Preserve Input Sample Rate.



16 Ops per Input-Output Sample
Replaces 400-Tap Requiring 400 Ops per Input Output

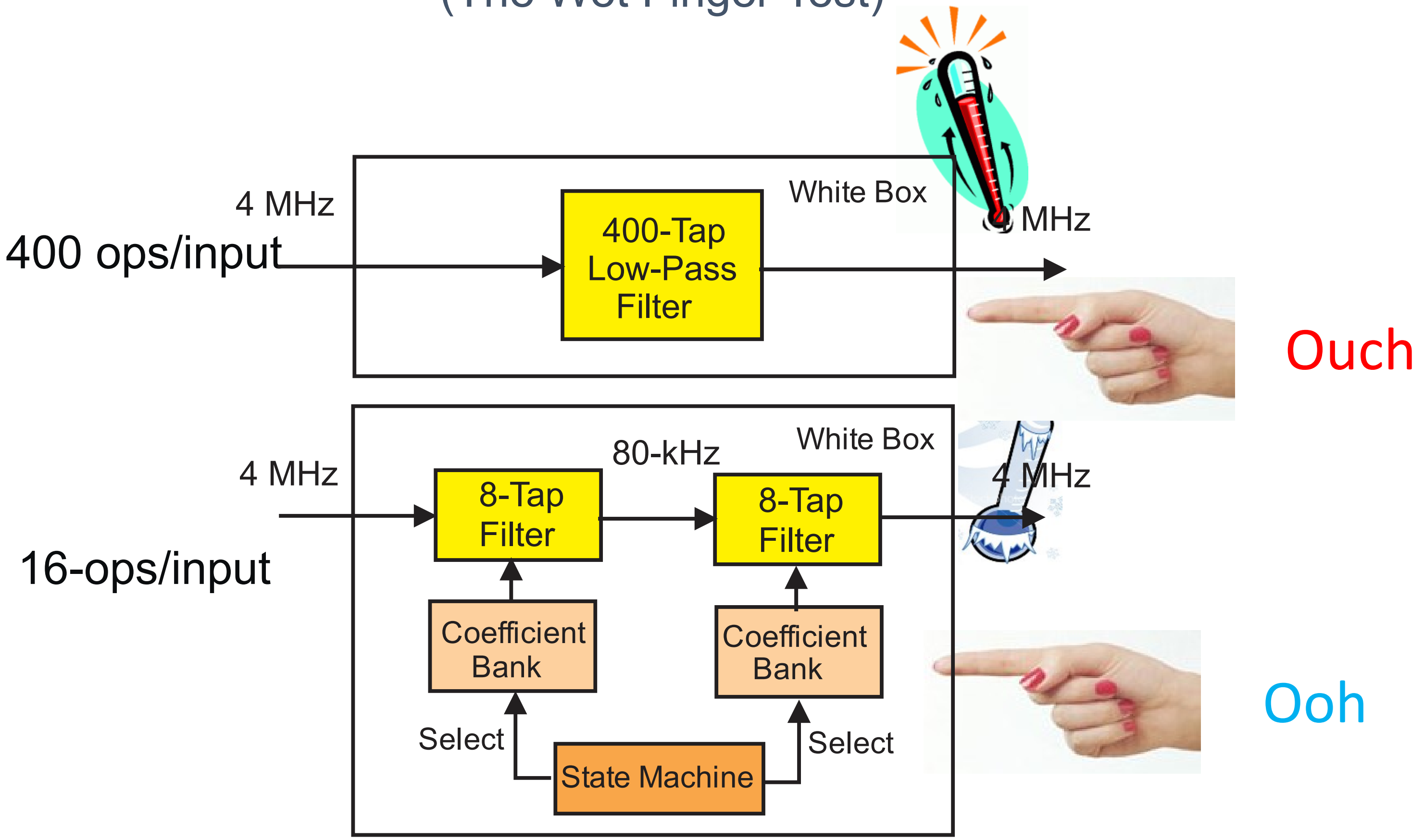
Efficient Polyphase Filter

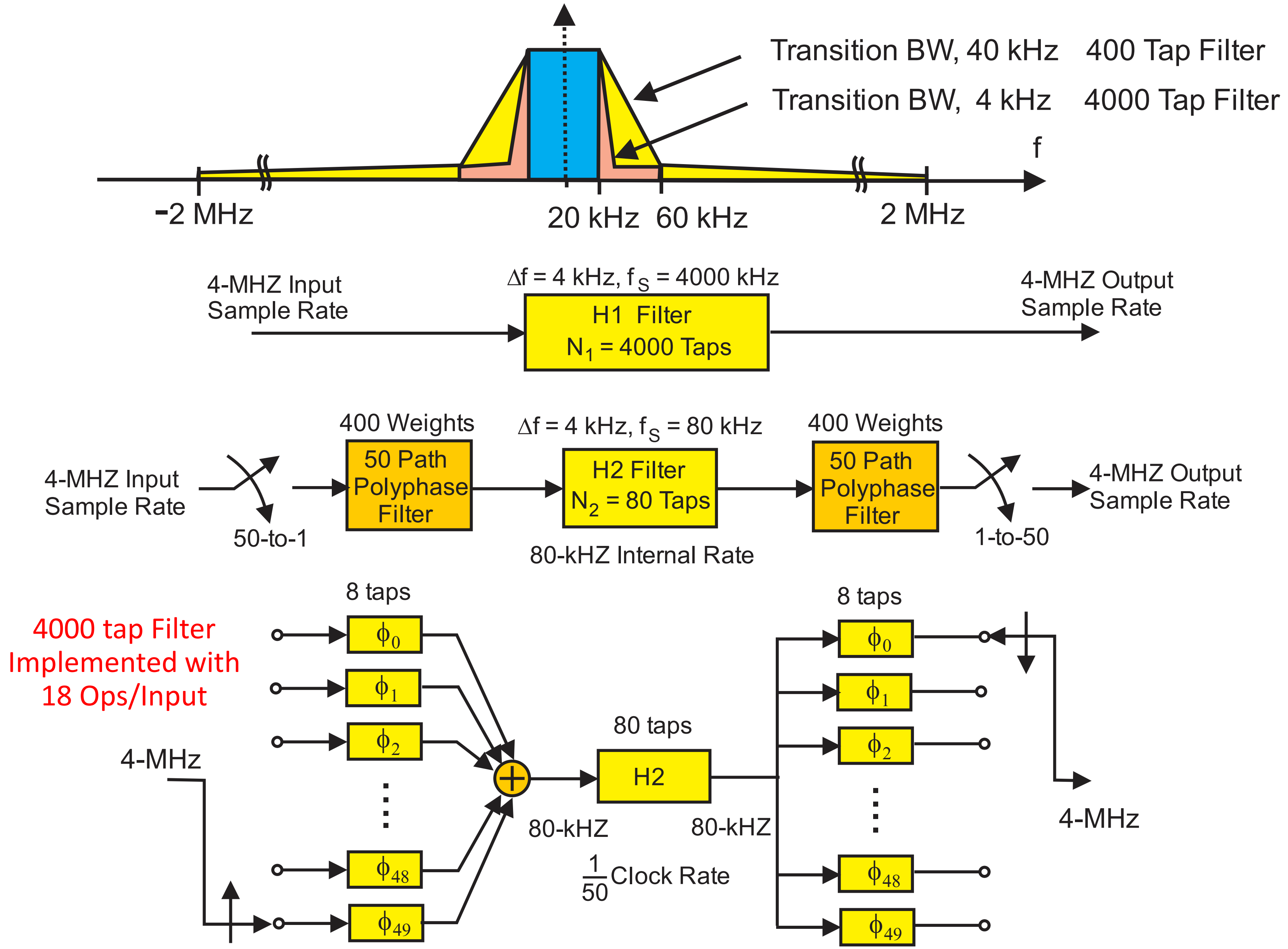


16 Ops per Input-Output Sample Replaces
400-Tap Filter Requiring 400 Ops per Input Output Sample

Different Processes in Two Boxes:
How can you tell which is which from outside box?

(The Wet Finger Test)

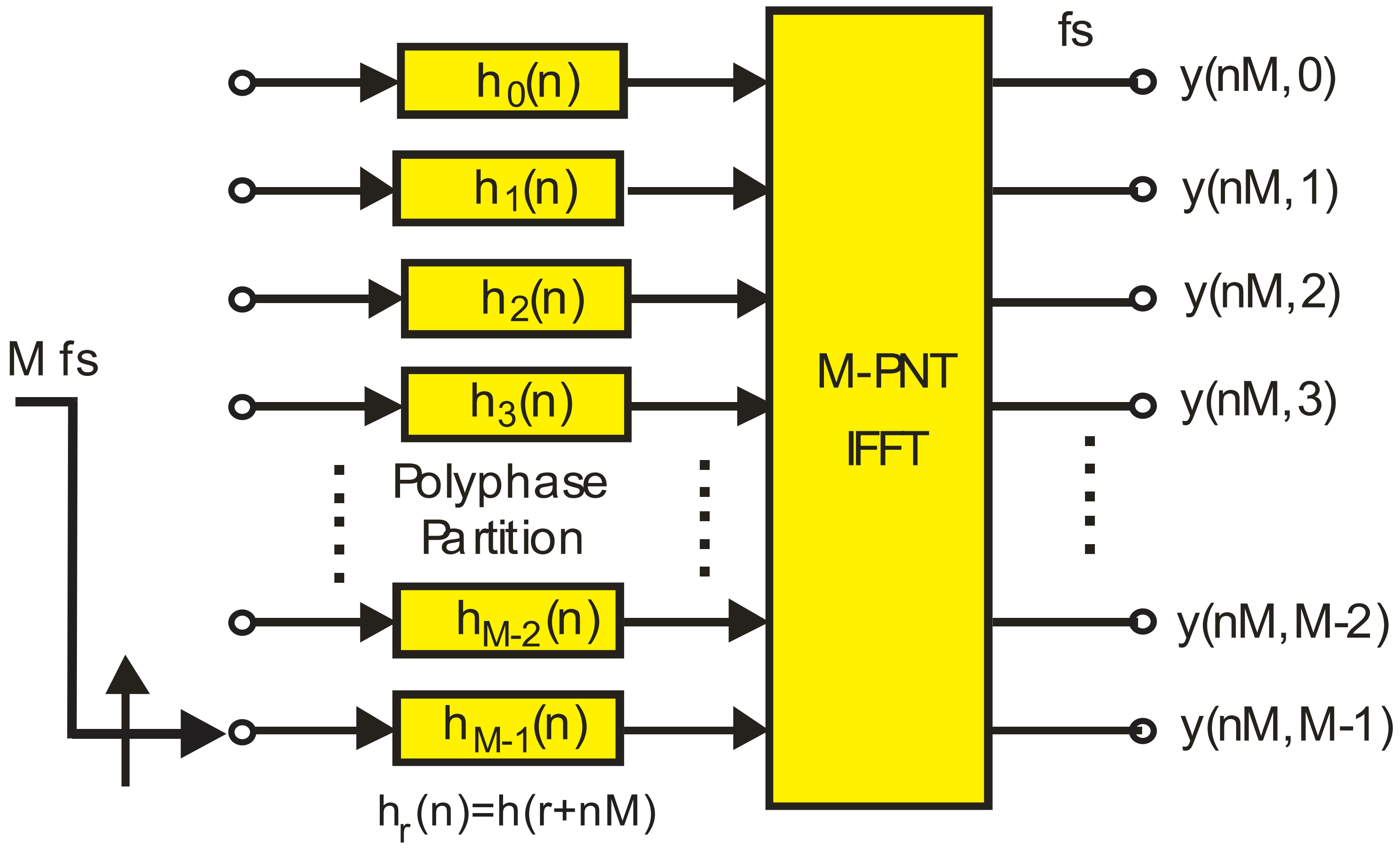




Inner Filter:
1/M Length Reduction,
1/M Clock Reduction,
$1/M^2$ Workload Reduction

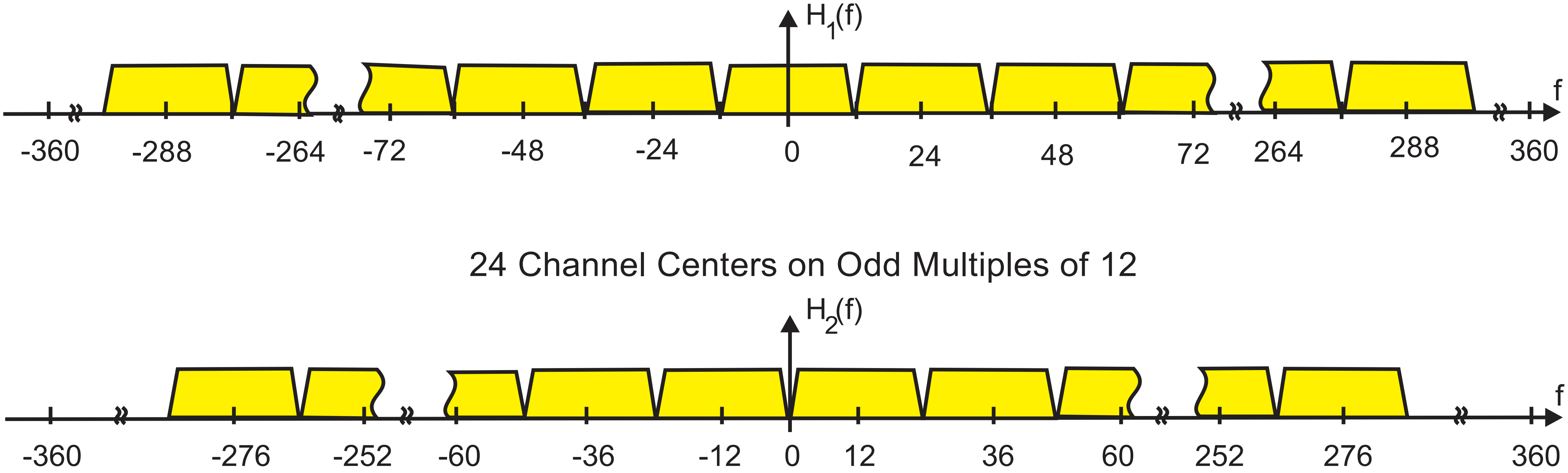
Standard M-Path Polyphase Analysis Channelizer

Channel Spacing from IFFT
Channel Bandwidth from Filter Prototype
Output Sample Rate for Input Commutator



Conventional Channelizer Center Frequencies Match Frequencies of M-Point IFFT, the integer Multiples of fs/M, k·fs/M
We Identify these Frequencies as the Even Multiples of fs/(2M), (2k)·fs/(2M)

25 Channel Centers on Even Multiples of 12



24 Channel Centers on Odd Multiples of 12

Offset Channelizer Center Frequencies are Midway Between Frequencies of M-Point IFFT, the Integer plus 1/2 Multiples of fs/M, (k+0.5)·fs/M
We Identify these Frequencies as the Odd Multiples of fs/(2M), (2k+1)·fs/(2M)

Receiver Channelizer Specifications

- Channel Spacing: 24.0 MHz
- Channel Bandwidth: 23.5 MHz
- Channel Sample Rate: 24.0 MHz
- Transition Bandwidth: 0.5 MHz
- Number of Channels: 24
- IFFT Size: 30
- Input Sample Rate: 720 MHz
- In-Band Ripple: 0.1 dB
- Stop band Attenuation: 50 dB
- Linear Phase FIR Filter

$$N_{Taps} \cong \frac{fs}{\Delta f} \frac{atten(dB)}{20}$$

$$= \frac{720}{0.5} \frac{50}{20}$$

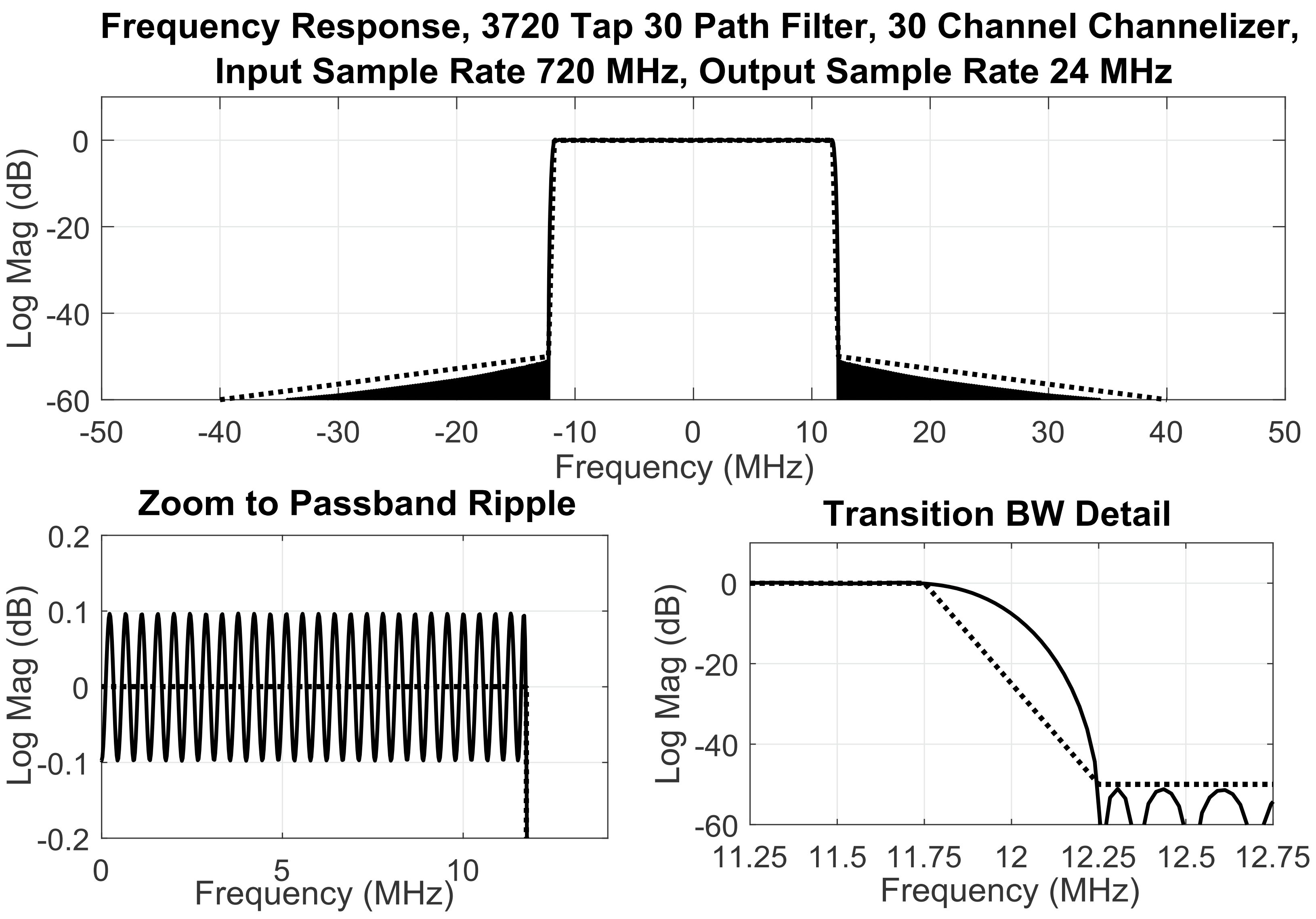
$$= 1440 \cdot 2.5 = 3600\ Taps$$

$$Actual = 3720\ Taps$$

Due to 1/f stopband slope

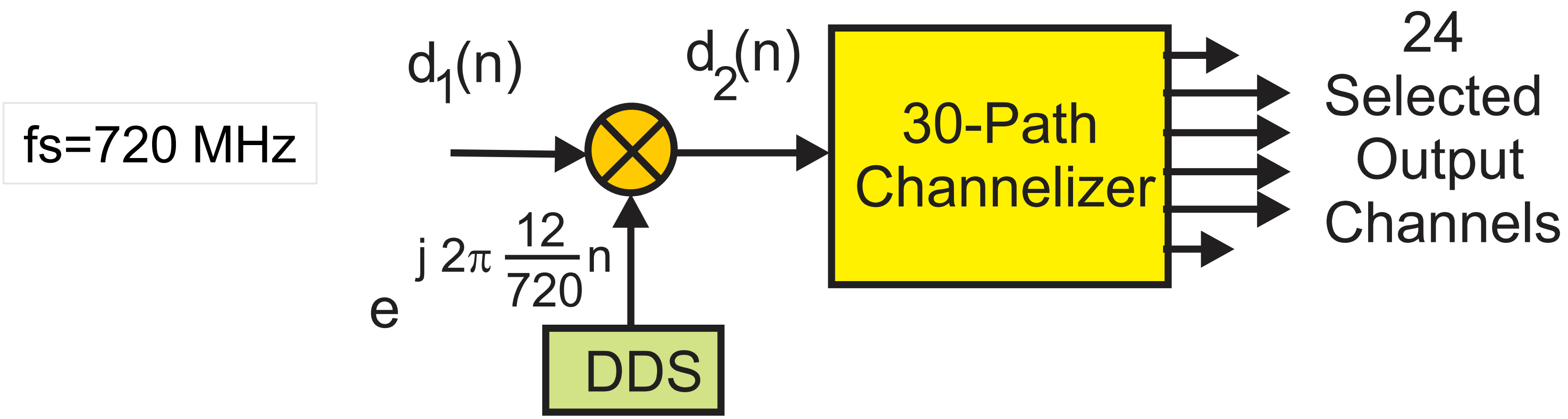
Number of Taps in Prototype
FIR Filter: 3720 Taps

Frequency Response of Prototype Filter
In 30-Path Filter, 124 Taps per path at 24 MHz Sample Rate



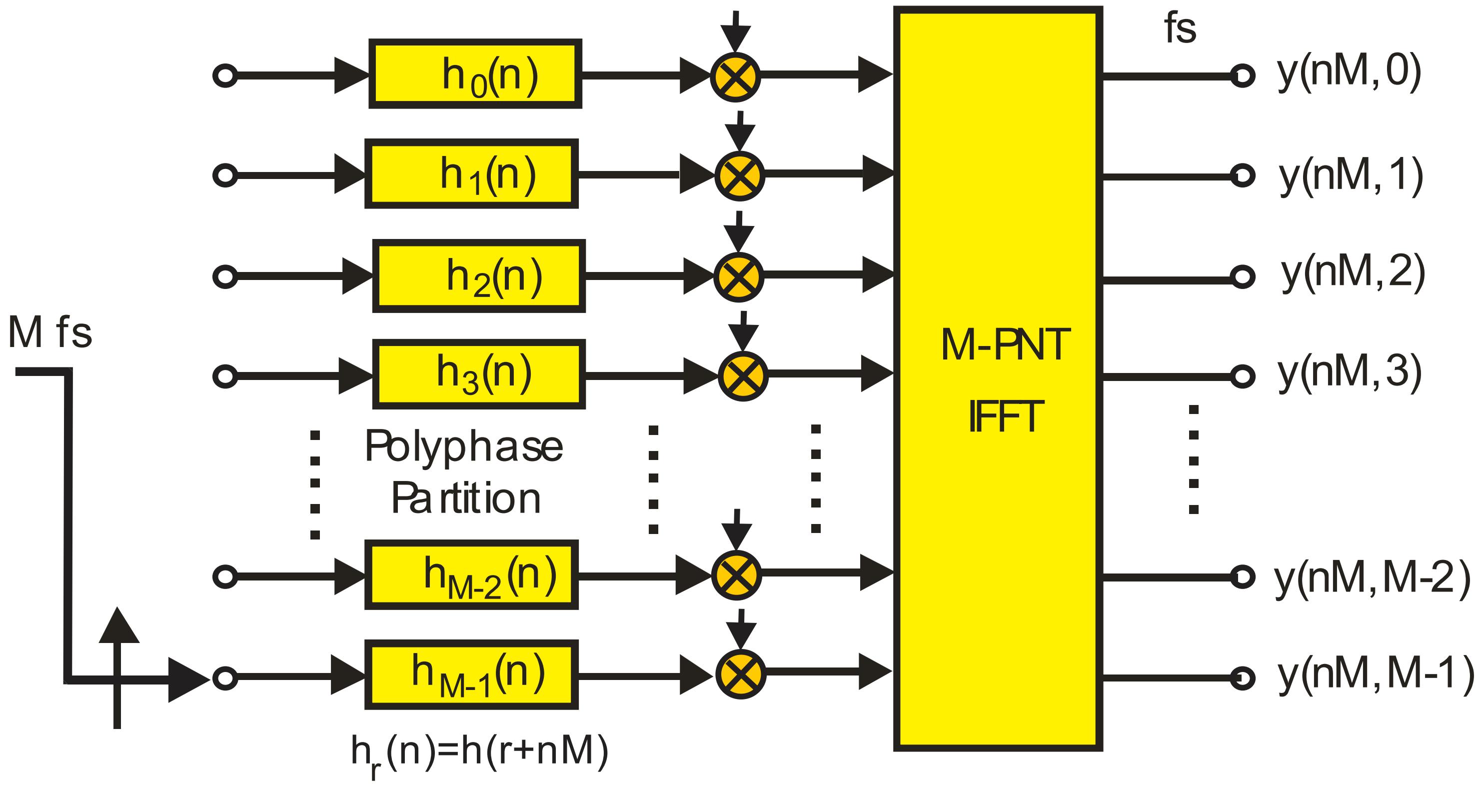
First Option to Convert Channelizer from Even Indexed Bin Centers to Odd Indexed Bin Centers Shift Input Spectra to Filter Spectral Locations

12 MHz Frequency Offset Heterodyne Operating at 720 MHz To obtain Odd Indexed Frequency Bins

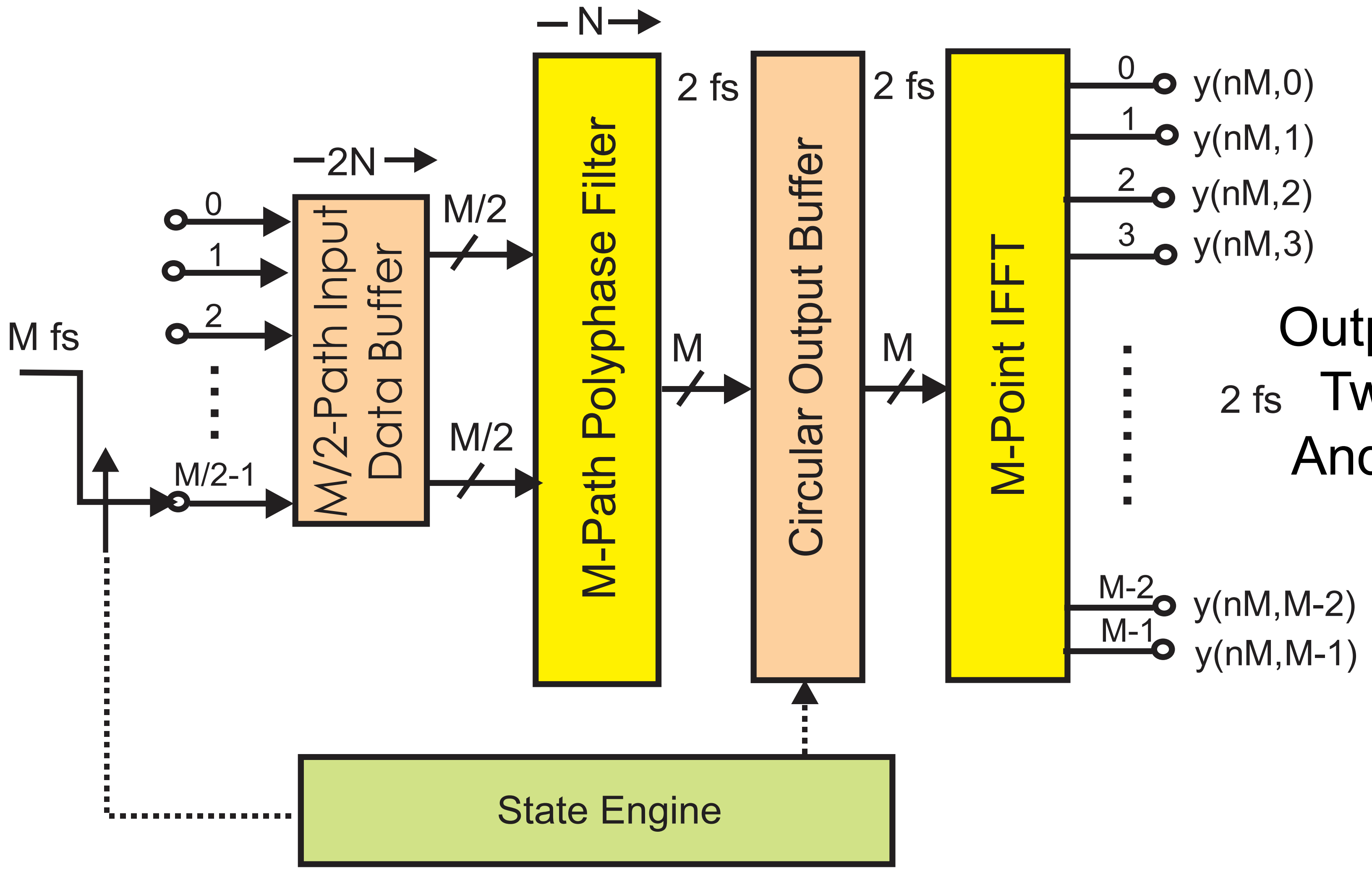


Second Option to Convert Channelizer from
Even Indexed Bin Centers to Odd Indexed Bin Centers
Shift Filter Spectra to Signal Spectral Locations

12 MHz Frequency Offset Heterodyne Embedded in M-Path Filter Arms
Operating at 24 MHz Sample Rate To obtain Odd Indexed Frequency Bins



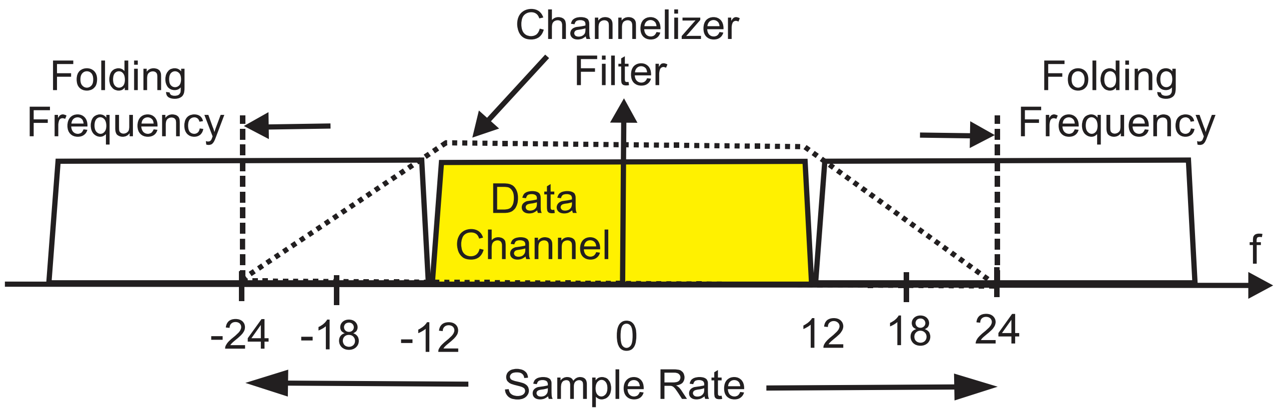
(M/2)-to-1 Downsampled Non-Maximally Decimated M-Path Filter Doubles Output Sample Rate

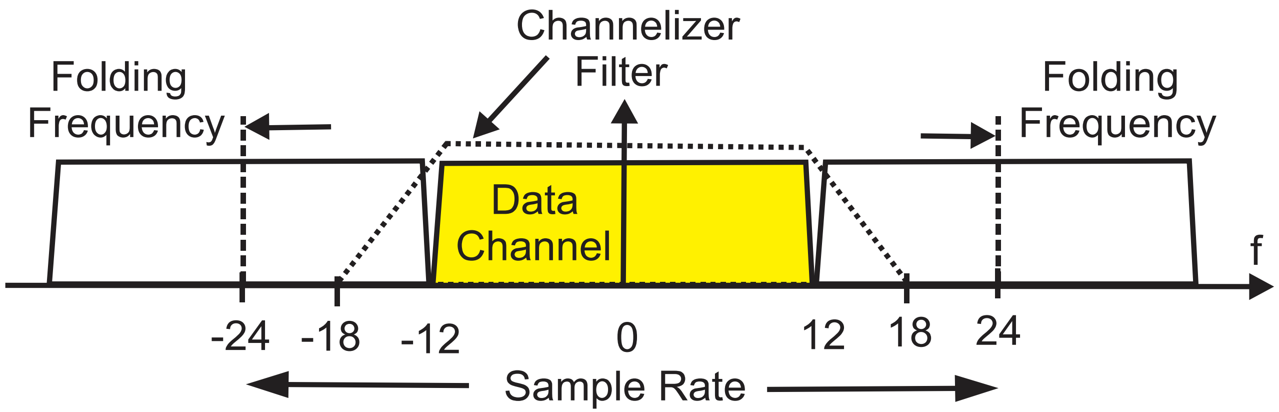


2 fs

Output sample Rate is now
Twice Channel Spacing
And is Twice Channel BW

With Increased Output Sample Rate we can Increase Transition Bandwidth



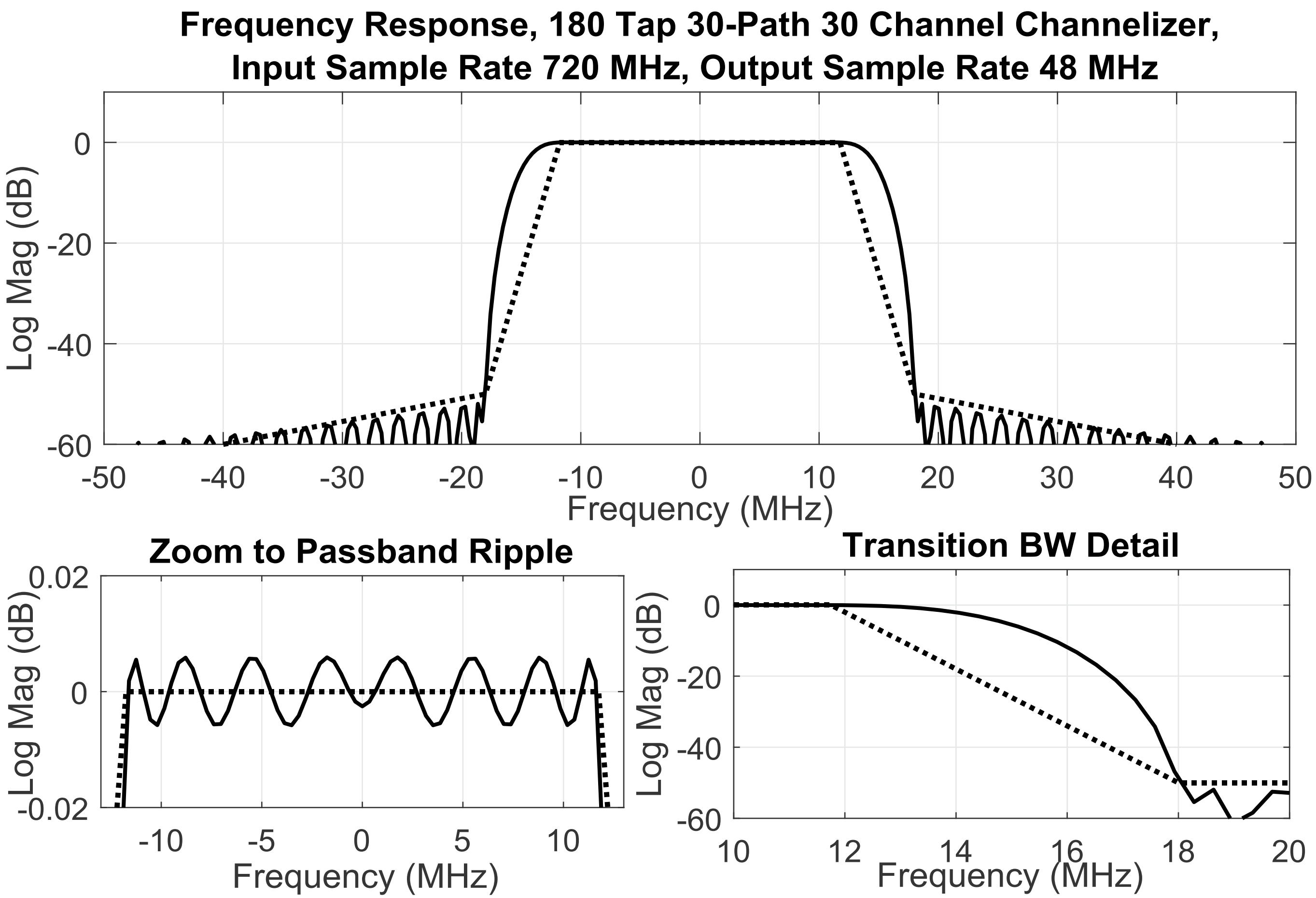


Original Transition BW was 0.5 MHz at 24 MHz Sample Rate

One Option is Change Transition BW to 12 MHz Which Would Reduce Filter Length by a Factor of 24

Second Option is Change Transition BW to 6 MHz Which Would let us see Stopband Level and Reduce Filter Length by a Factor of 12

Frequency Response of Prototype Filter
In 30-Path Filter, 6 Taps per path at 48 MHz Sample Rate

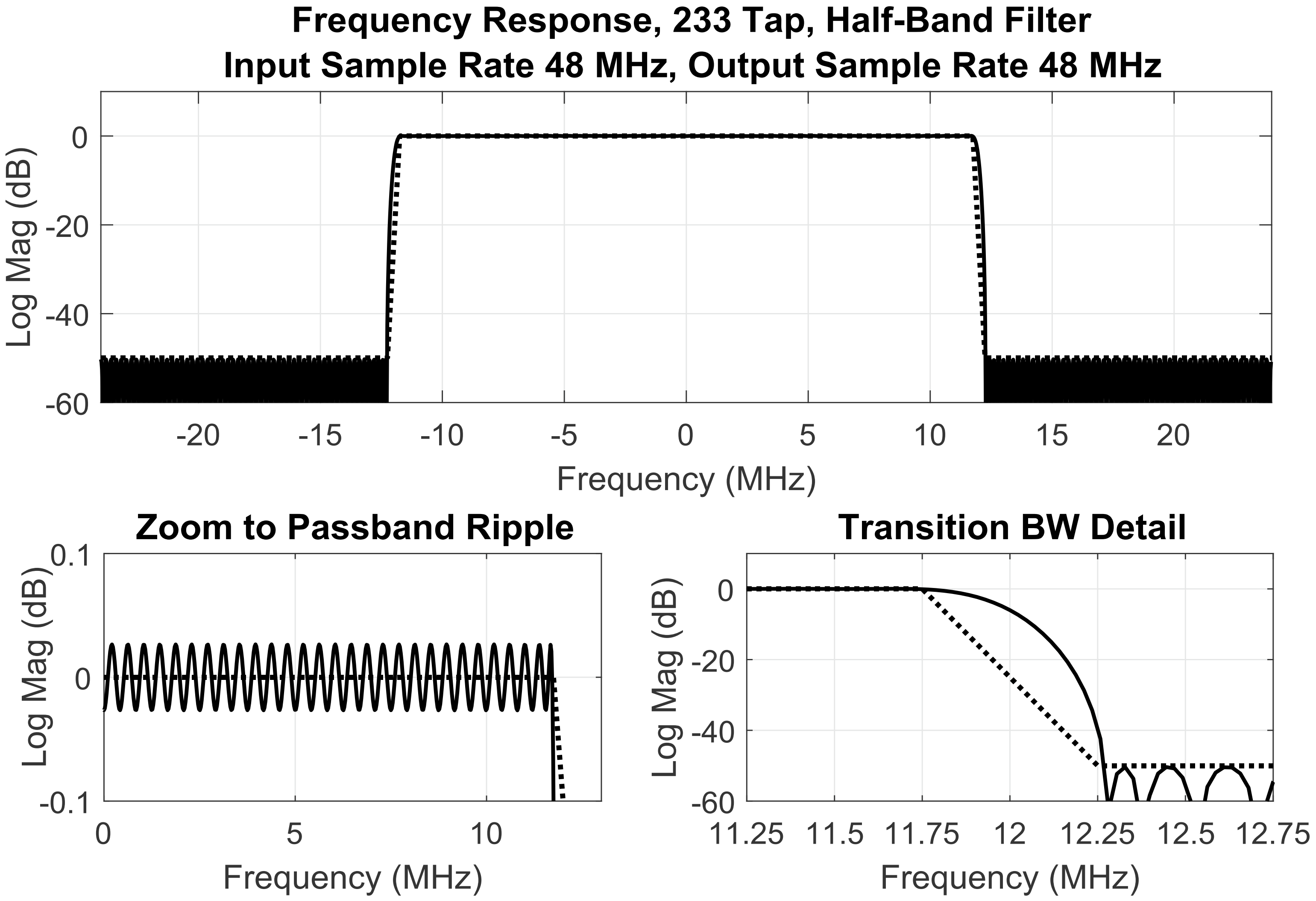


Significant Workload Reduction of 30-Path Filter

Originally 3720 Taps
124 Taps/Path

Now 180 Taps
6 Taps/Path

Following 30-Path Channelizer, we can Reduce Transition BW and Sample Rate with a Cascade Half Band FIR Filter on Each of the 24 Output Channels



116 Tap Delay in FIR Filter

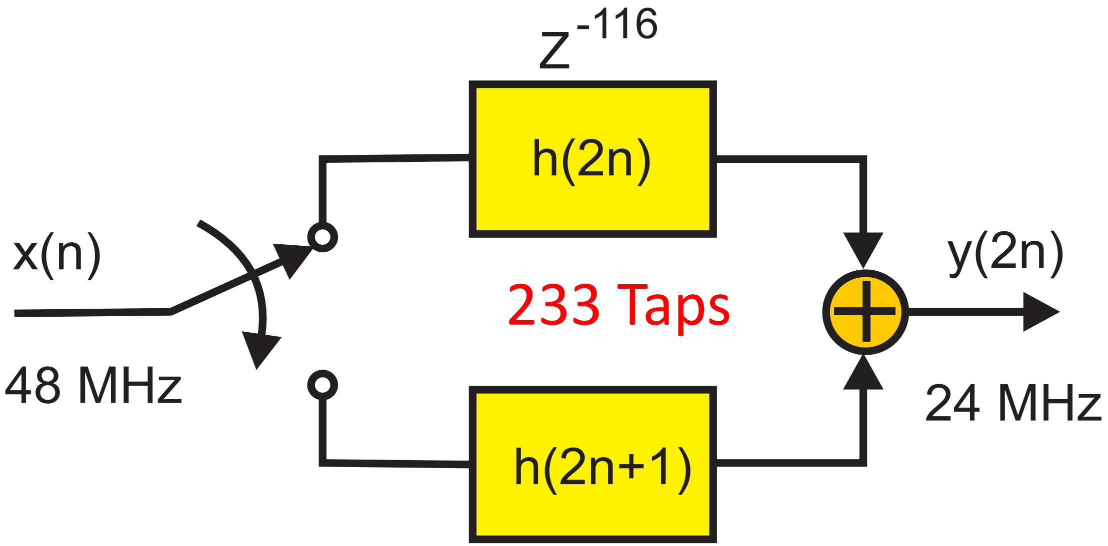
233 Taps have a Workload that is Larger than each path of original 30-Path Channelizer (124 Taps)

We are not Finished Shaving Workload

Two-Path Implementation of Half Band Filter Reduces Sample Rate 2-to-1 While Reducing BW

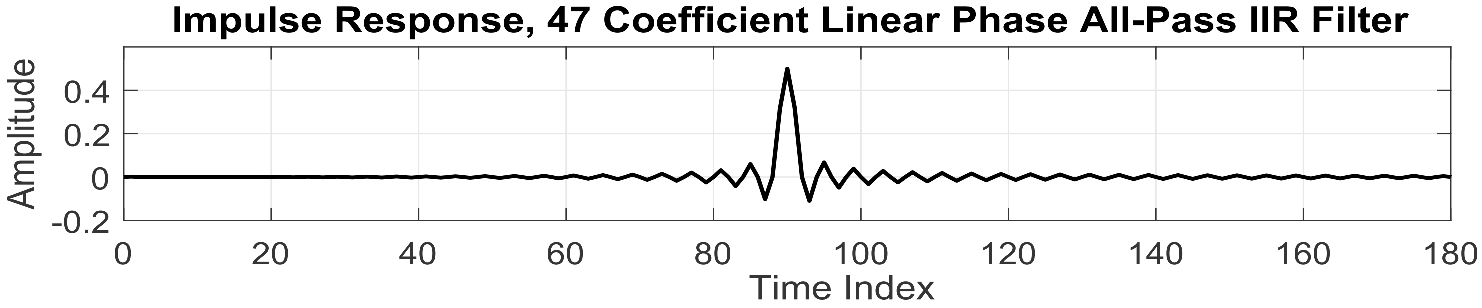
Half of Weights in True Half Band Filter are Zero Upper Path has one Non Zero Weight

Lower Path has Even Symmetric Weights can Fold Filter and Operate Lower Path with 58 Multiplies per Output Sample at 24 MHz Rate

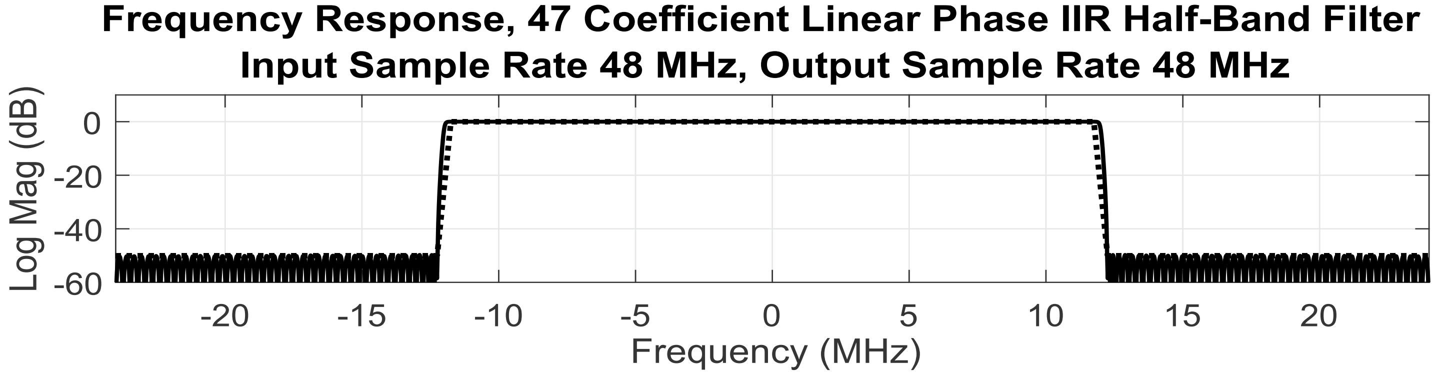


116 Odd Indexed
Even Symmetric Coefficients

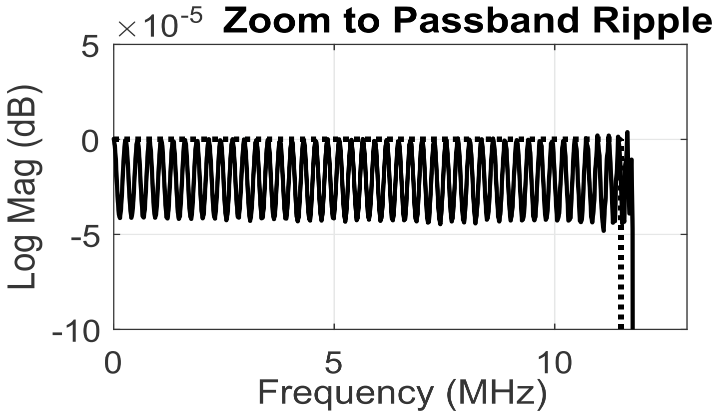
Following Channelizer, we can Reduce Transition BW and Sample Rate with a Cascade Half Band Linear Phase IIR Filter on Each of the 24 Output Channels

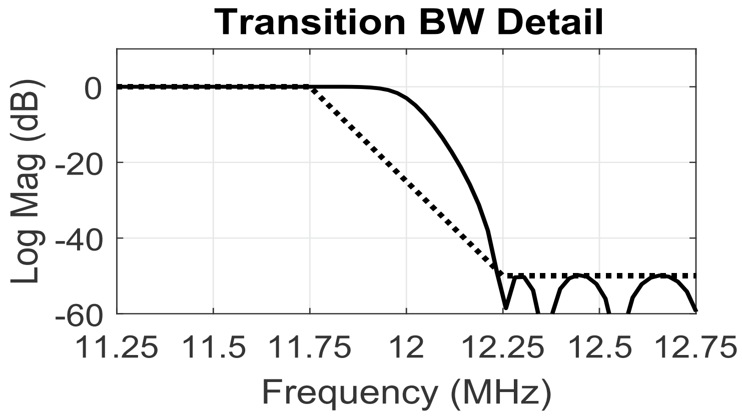


94 Tap Delay
in IIR Filter

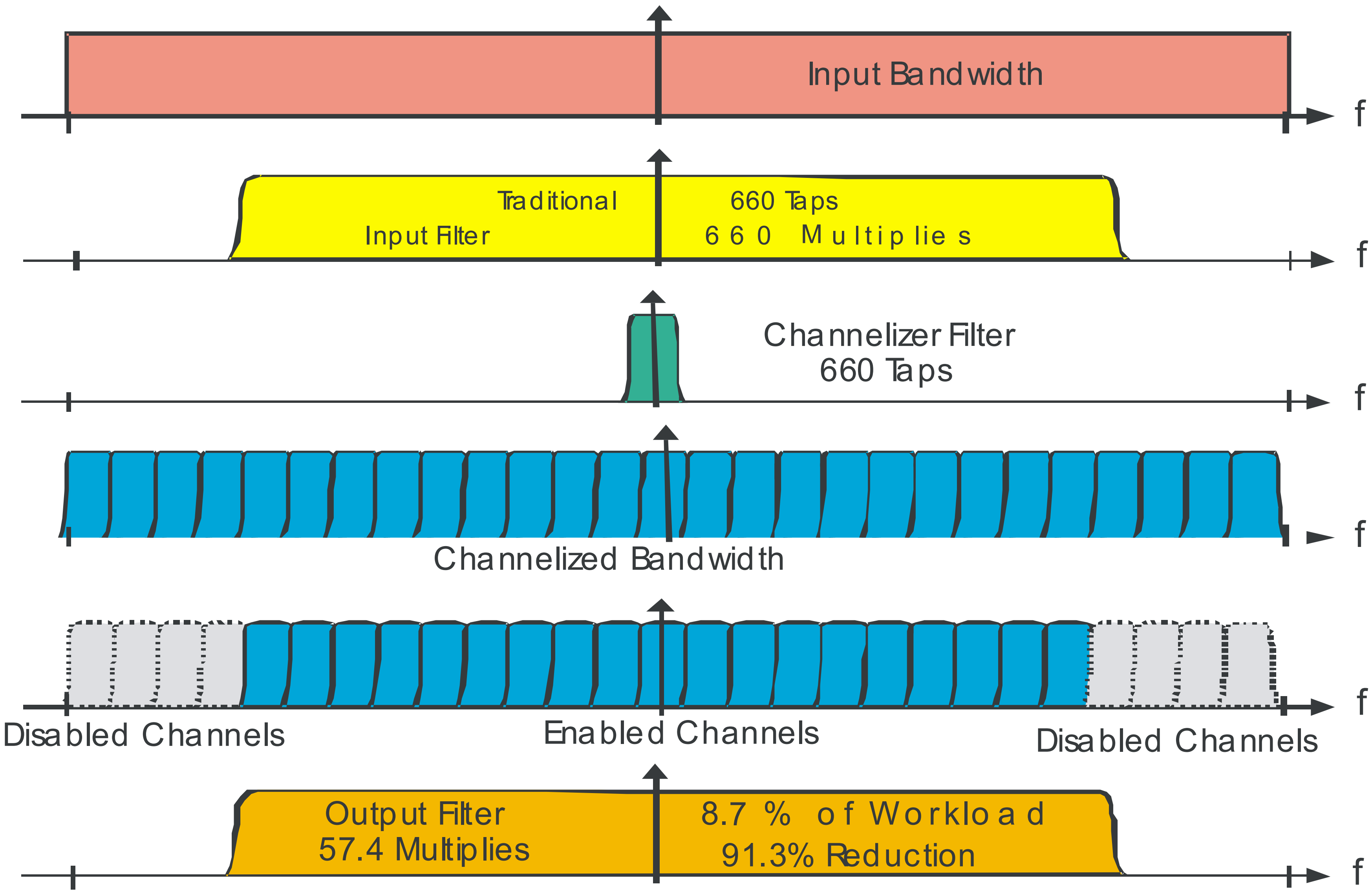


Lower Path of Linear Phase
Recursive 2-Path Filter
Operates with
47 Multiplies per Output
Sample at 24 MHz Rate

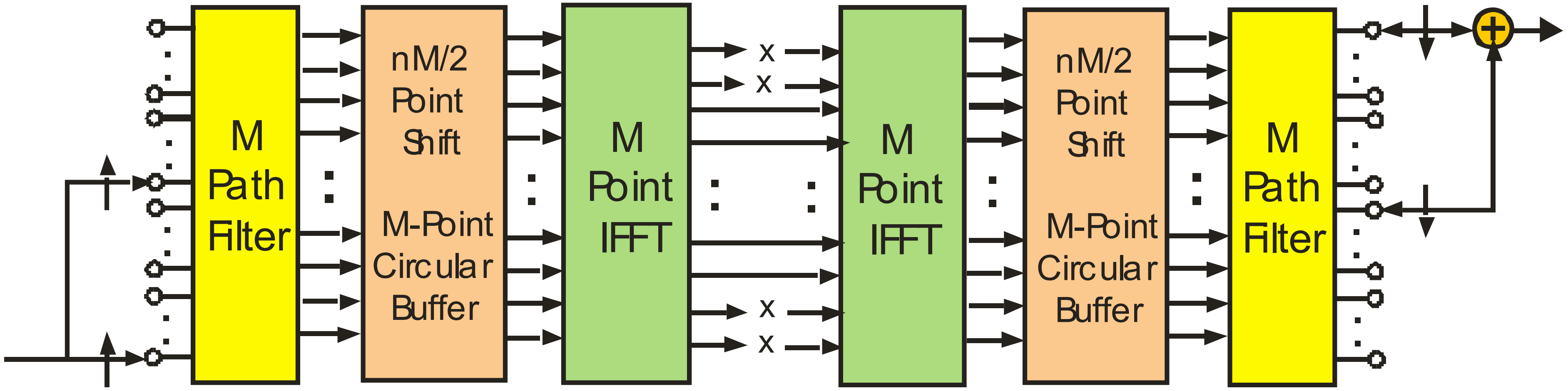




Compare Traditional Tapped Delay Line Filter Implementation to Polyphase Filter Implementation



Frequency Domain Filtering With Cascade M/2-to-1 Analysis and 1-to-M/2 Synthesis Channelizers



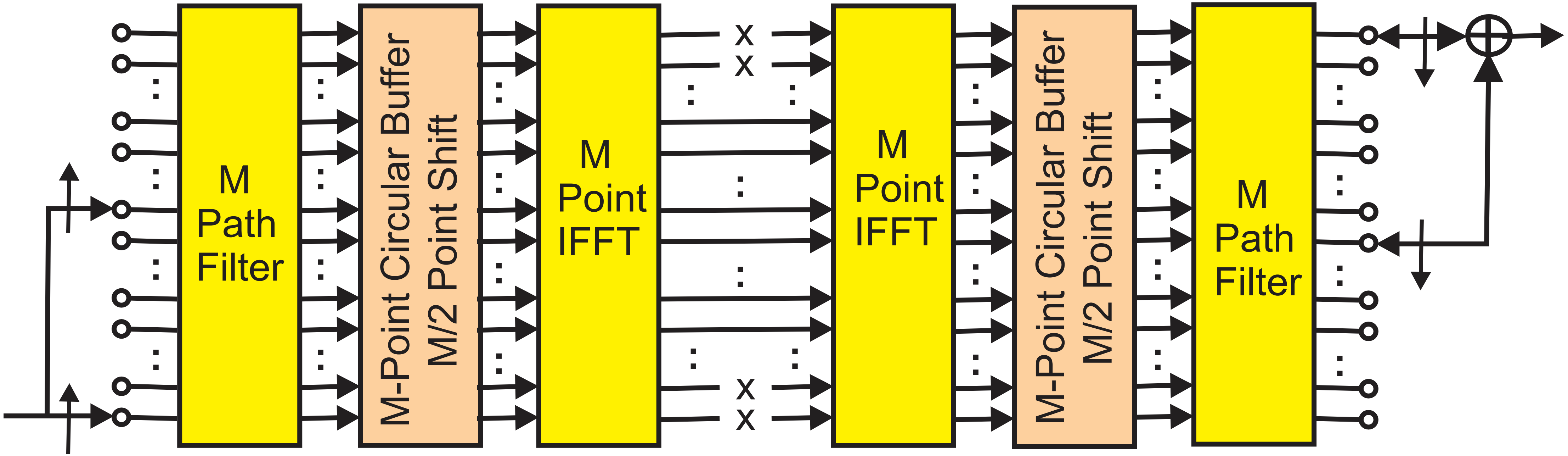
660 Tap Prototype Filter
60 Path Polyphase Partition
11 Coefficients per Path
Input Filter:
22 Operations Per Input Sample
22 Operations Per Output Sample

60 Point Good-Thomas Nested Winograd FFT
200 Real Multiples
Amortized over 15 Input Samples:
13.3 Operations per Input Sample
13.3 Operations per Output sample

For Input and Output Polyphase Filter and IFFT:
70.6 Operations per Input-Output Sample Pair

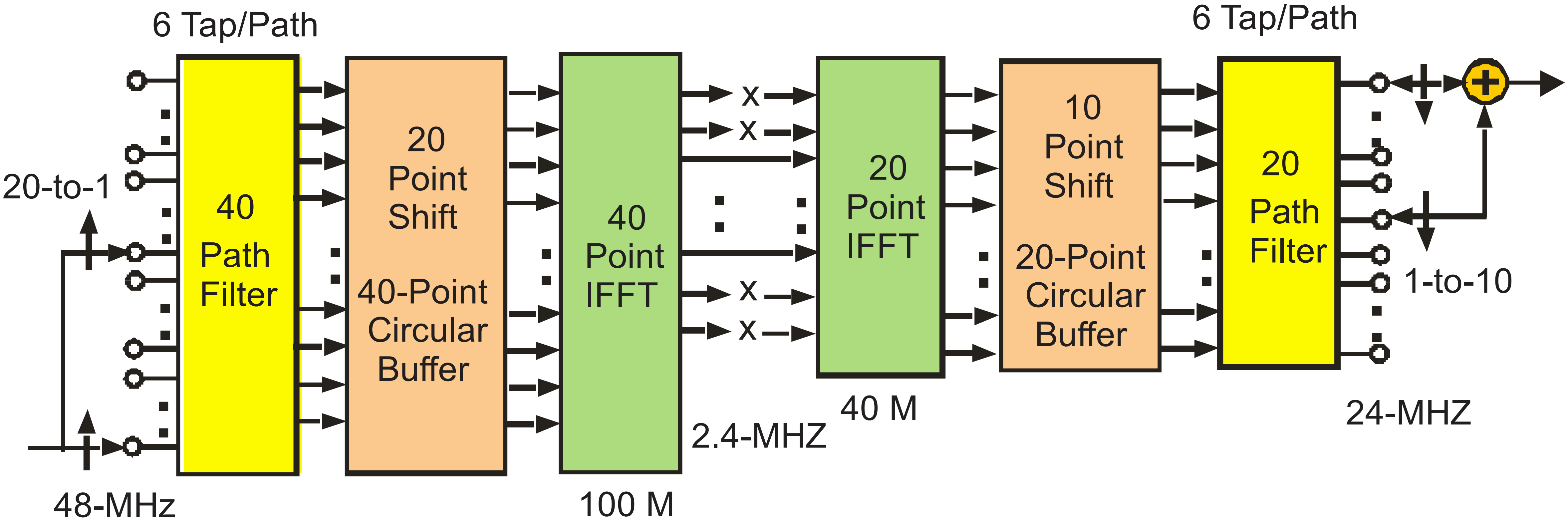
Workload: 70.6/660 = 10.7% of 660 Coefficient Tapped Delay Line Filter With same Frequency Response!

Following 30-Path Channelizer, we can Reduce Transition BW and Sample Rate with a Cascade 40-Path Analysis and Synthesis Channelizers on Each of the 24 Output Channels



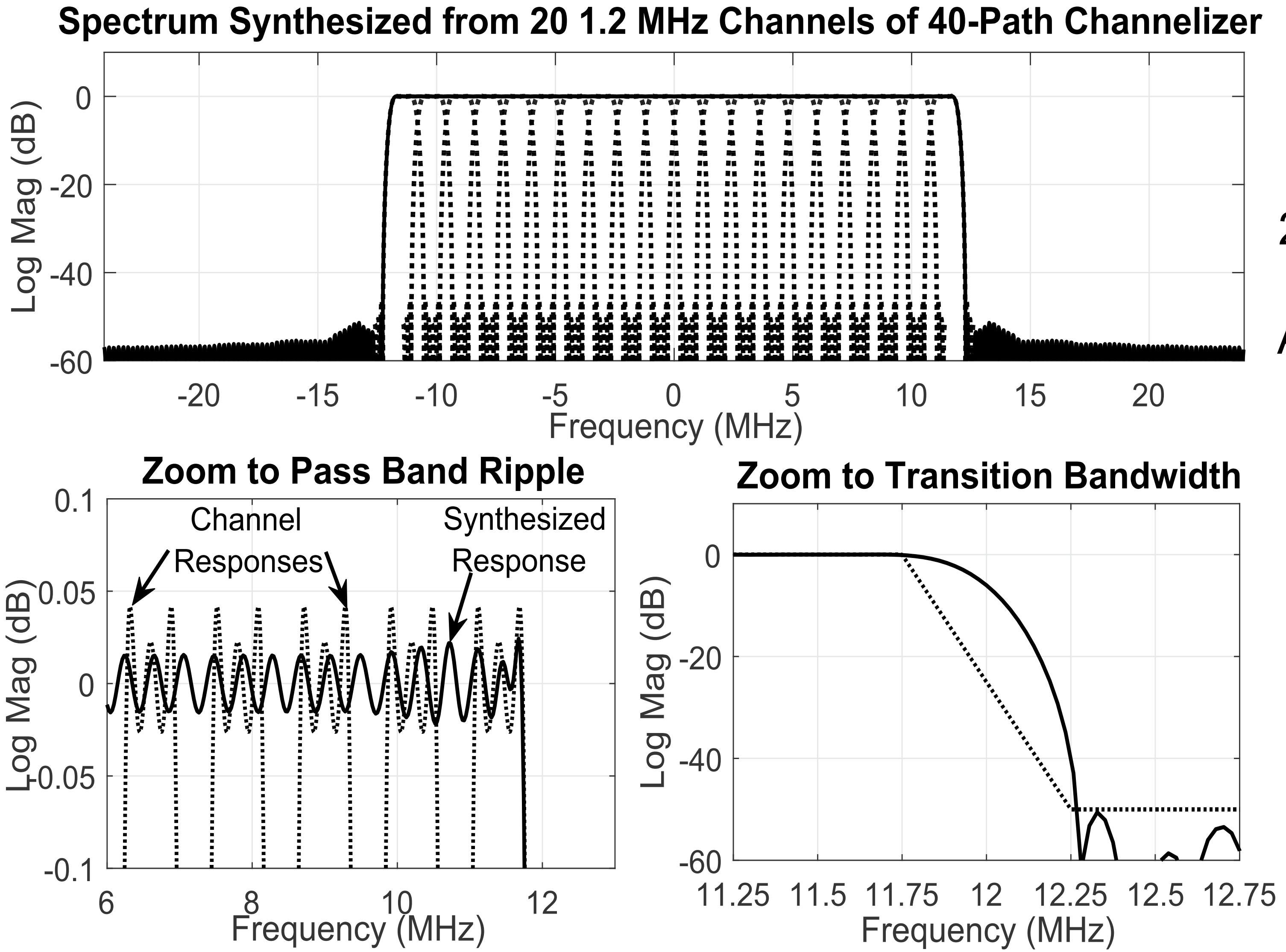
40 Path Filter, 6-taps per path
40 Point IFFT, 5*8 Good-Thomas, Winograd, 100 multiplies
Workload: 40-Path Filter,12-multiplies per input
40 Point IFFT 100/20, 5 Multiplies per 20 Inputs
34 multiplies per input output sample, Operating at 2.4 MHz Channelizer Rate

Following 30-Path Channelizer, we can Reduce Transition BW and Sample Rate with a Cascade 40-Path Analysis and 20 Path Synthesis Channelizers on Each of the 24 Output Channels



40 Path Filter, 6-taps per path
40 Point IFFT, 5*8 Good-Thomas, Winograd, 100 multiplies
Workload: 40-Path Filter,12-multiplies per input
40 Point IFFT 100/20, 5 Multiplies per 20 Inputs
25 multiplies per input output sample, with 20 Path and 20 Point Output Filter

Following 30-Path Channelizer, we can Reduce Transition BW and Sample Rate with a Cascade 40-Path Analysis and 20-Path Synthesis Half Band Super Filter on Each of the 24 Output Channels



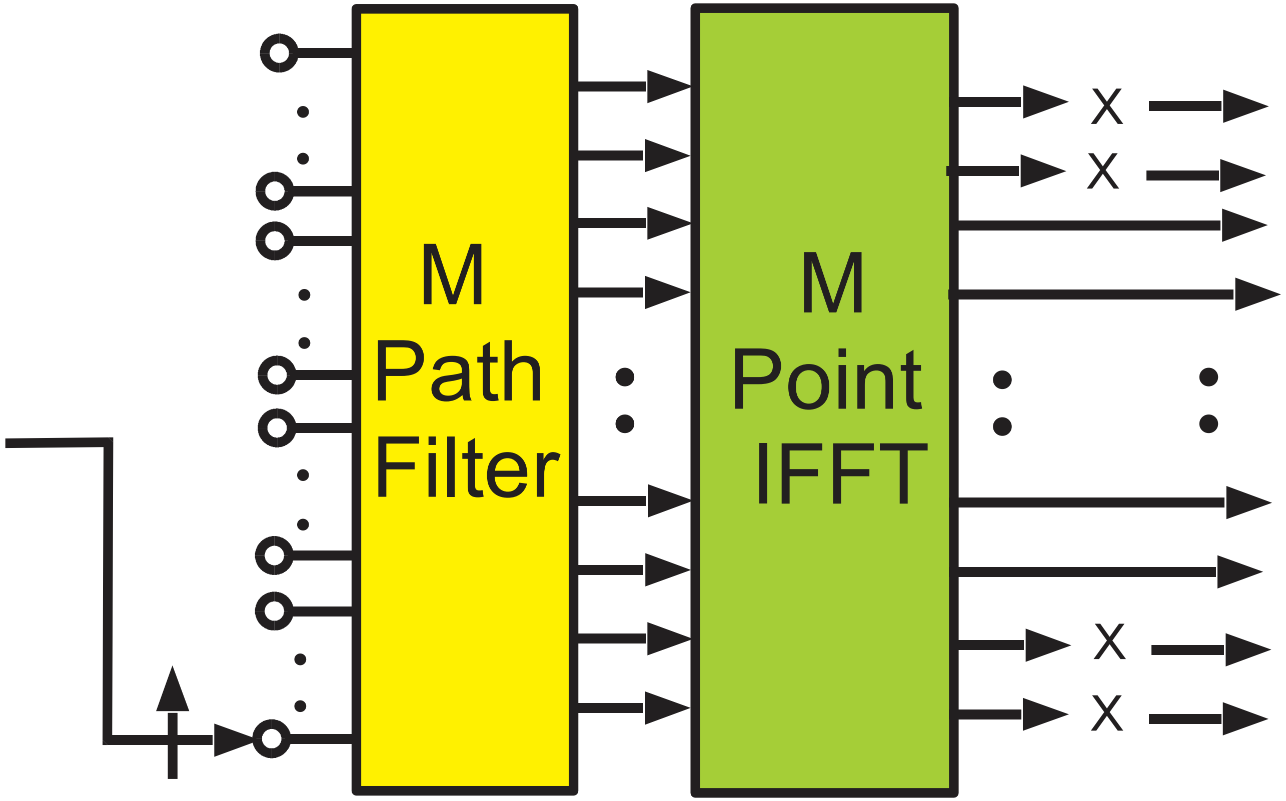
25 multiplies per input-output sample, with 40 Path Input Analysis and 20 Point Output Synthesis Channelizer Synthesized Super Filter

Lesson Learned in Multirate Channelizers

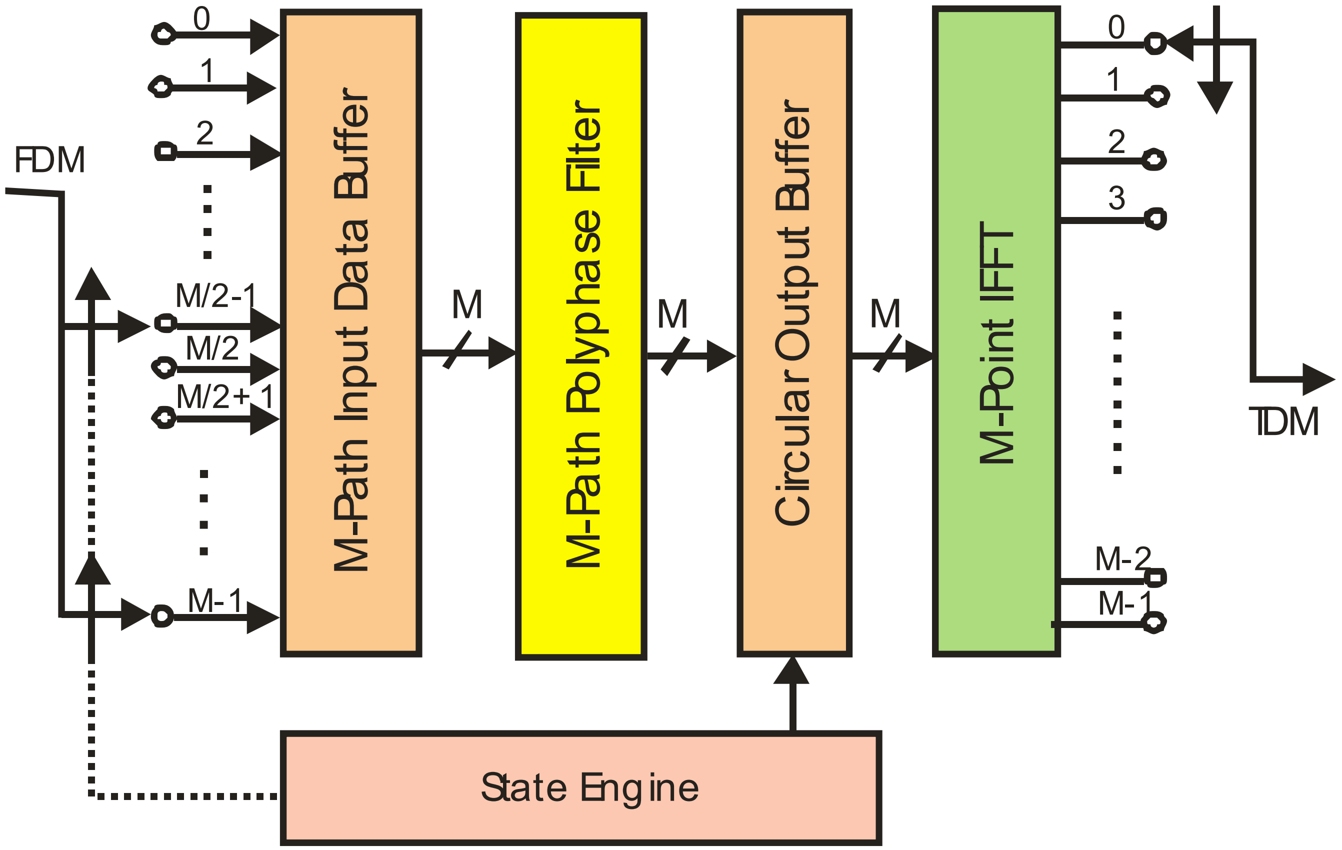
- Don't Design Filter with Narrow Transition BW and High Sample Rate.
- Solve Problem with Cascade of two Filters
  - First to Reduce BW and Sample Rate with Wide Transition BW
  - Second to Reduce Transition BW and
    Sample Rate at Reduced Sample Rate
- Design Signal Conditioning First Filter with
  Reduced number of taps and Wider Transition BW
- Design second Filter to operate at its Reduced Nyquist Sample Rate
- Reduce Sample Rate to Nyquist Rate with Non Maximally
  Decimated M-Path Polyphase Filter
- Design and Operate Second Filter at Reduced Rate
  with Reduced Number of Coefficients
- If Required, Increase Sample Rate with M-Path Polyphase Filter

Analysis and Synthesis Channelizers with Even and Odd Indexed Bin Centers

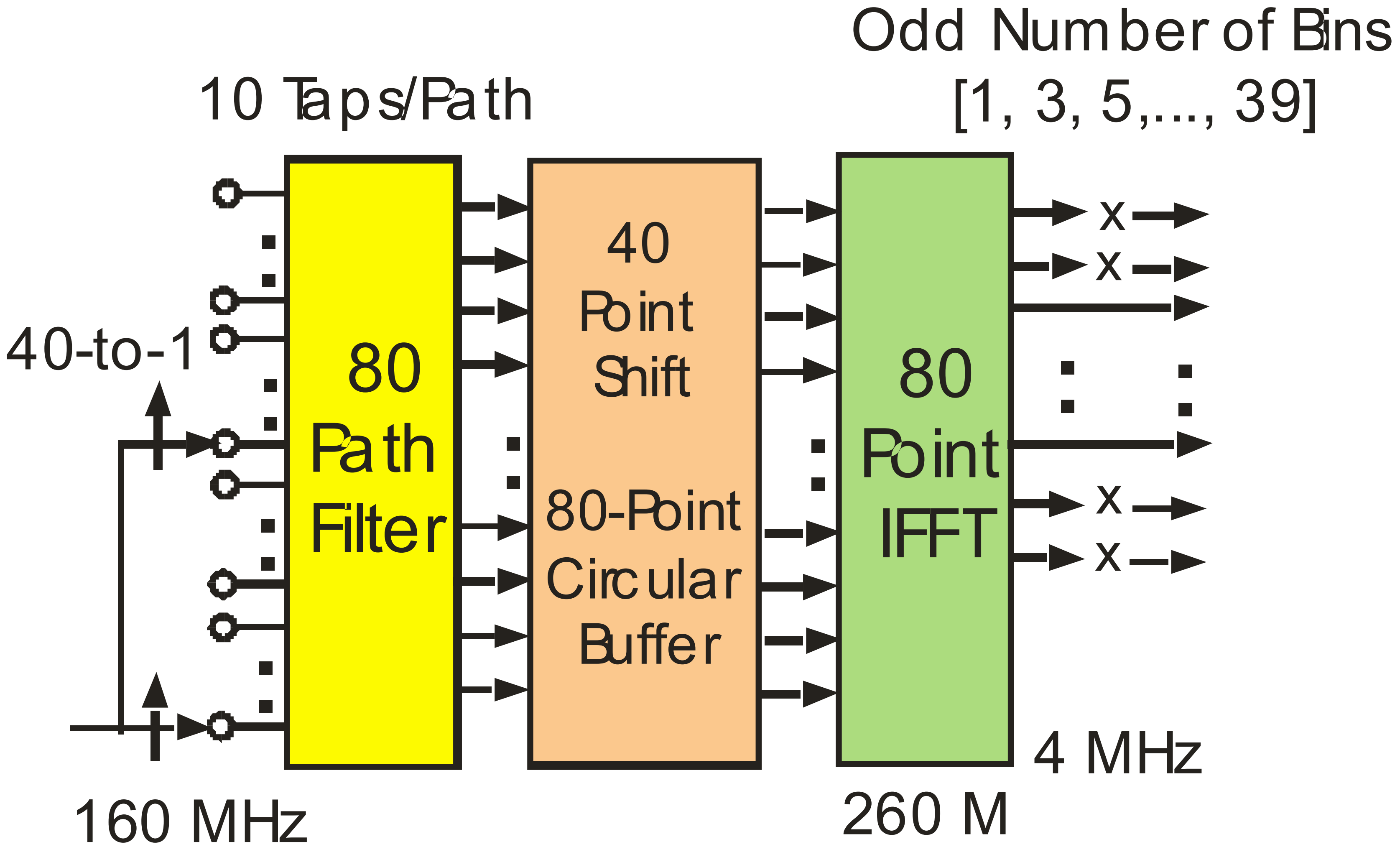
M-Path Analysis Filter Bank with M to 1 Down Sampling
Maximally Decimated Filter Bank



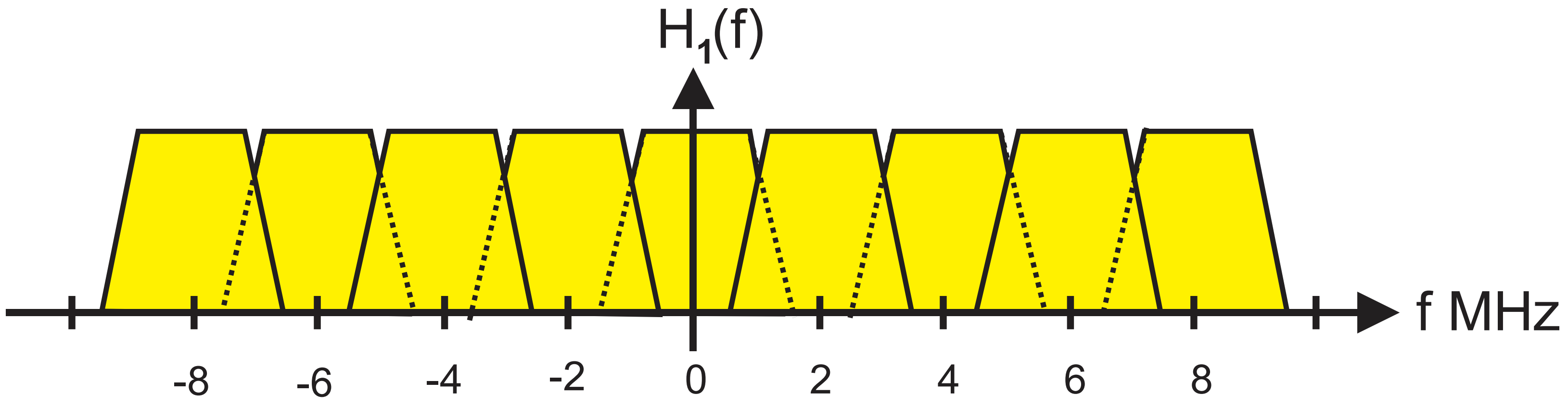
M-Path Analysis Filter Bank with M/2 to 1 Down Sampling
Non Maximally Decimated Perfect Reconstruction Filter Bank



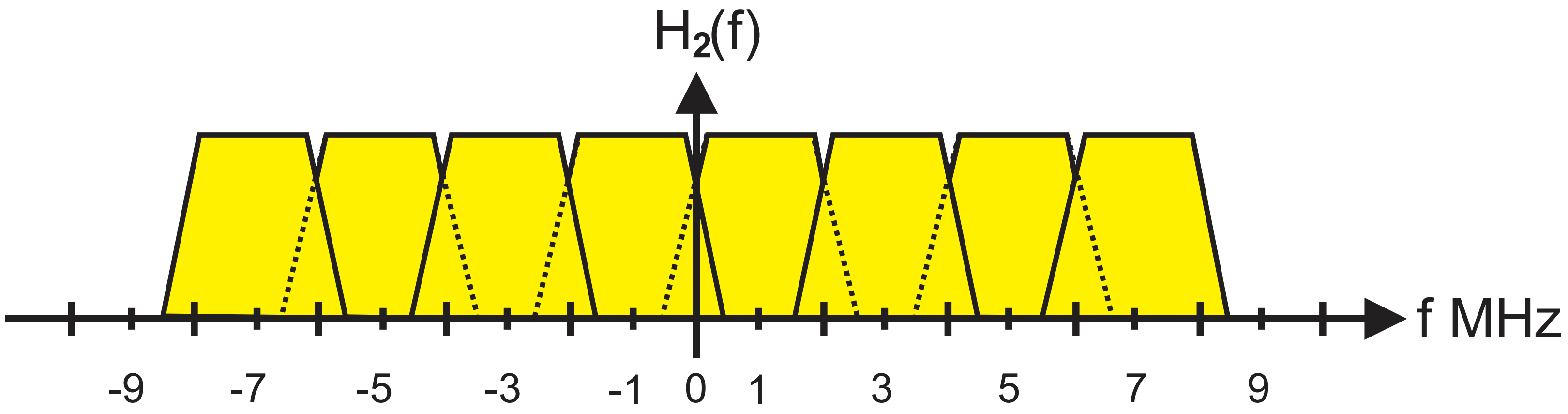
M-Path Analysis Filter Bank with M/2 to 1 Down Sampling
Non Maximally Decimated Perfect Reconstruction Filter Bank



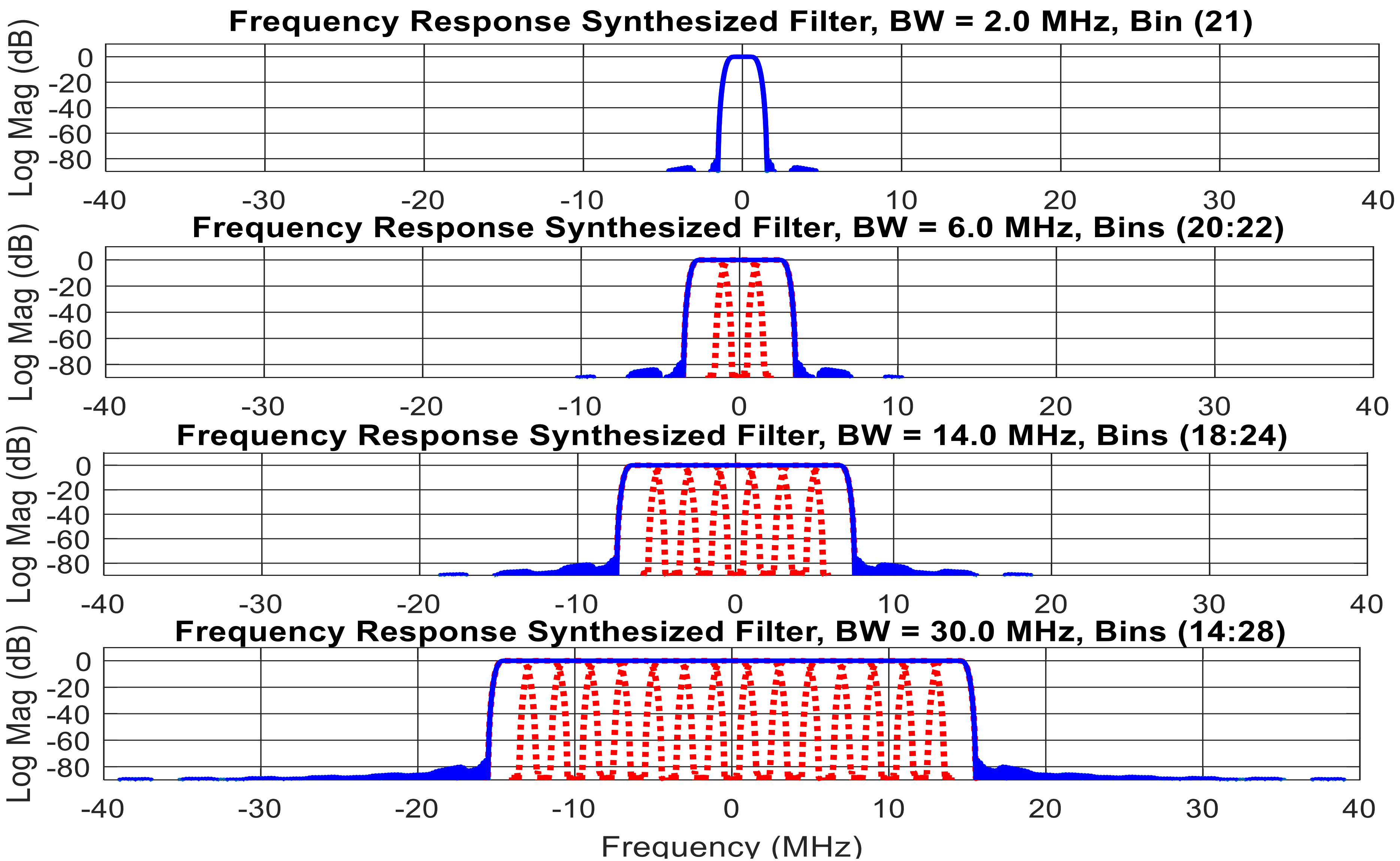
Even Indexed Channelizer



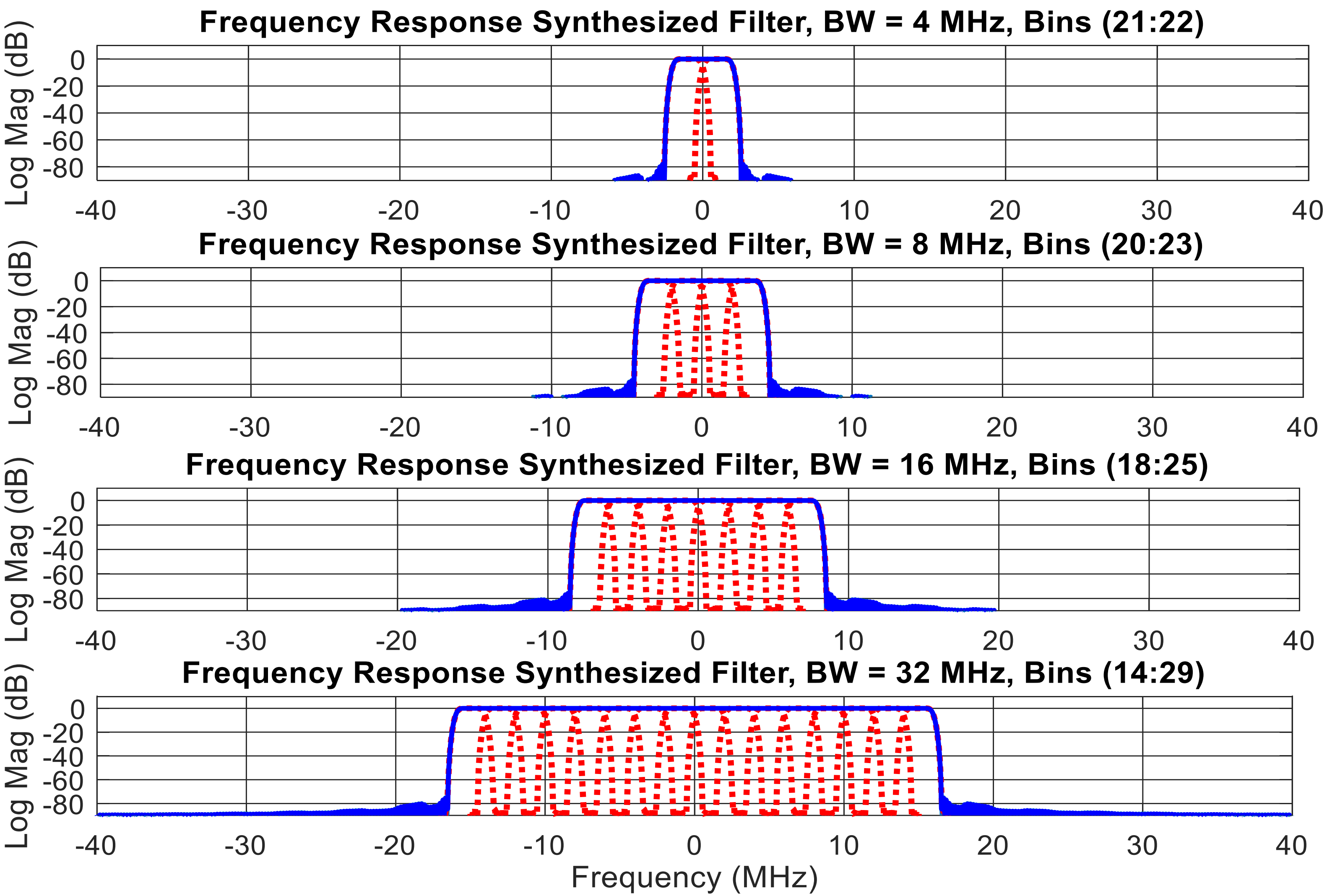
Odd Indexed Channelizer



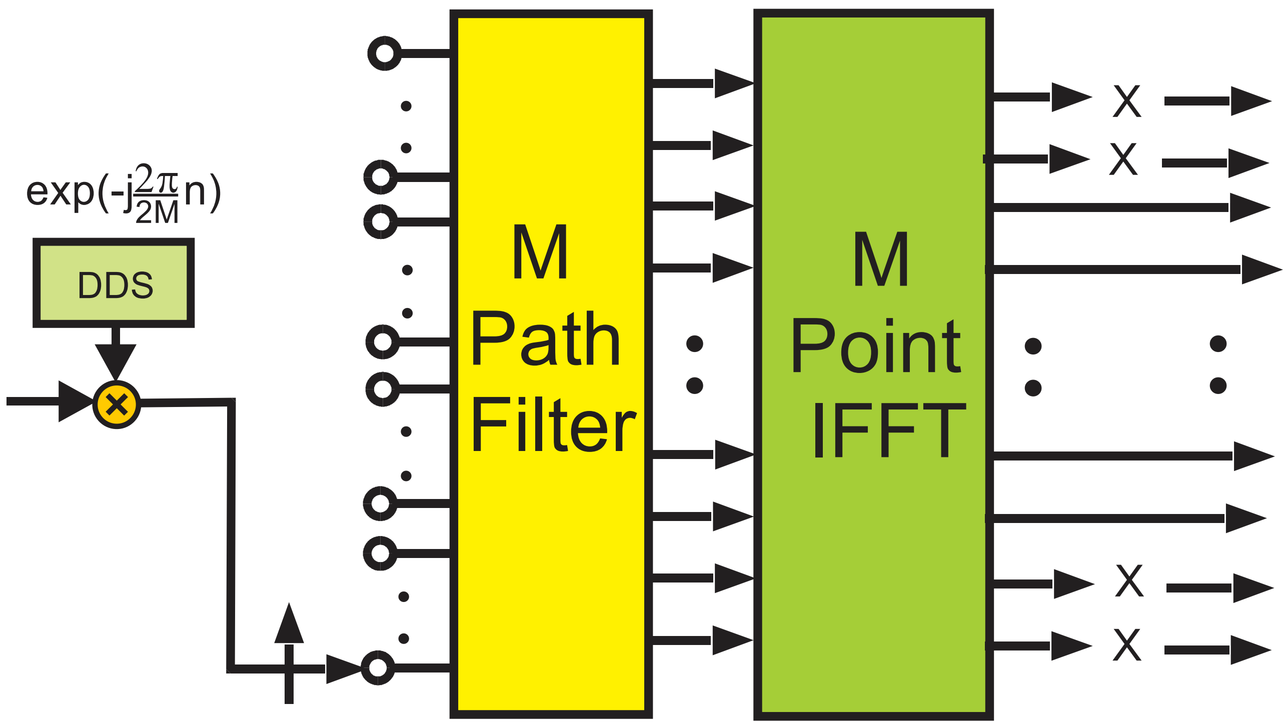
Center Frequencies: Match Roots of Z^N-1, N Roots of 1, exp(j 2π k/N)

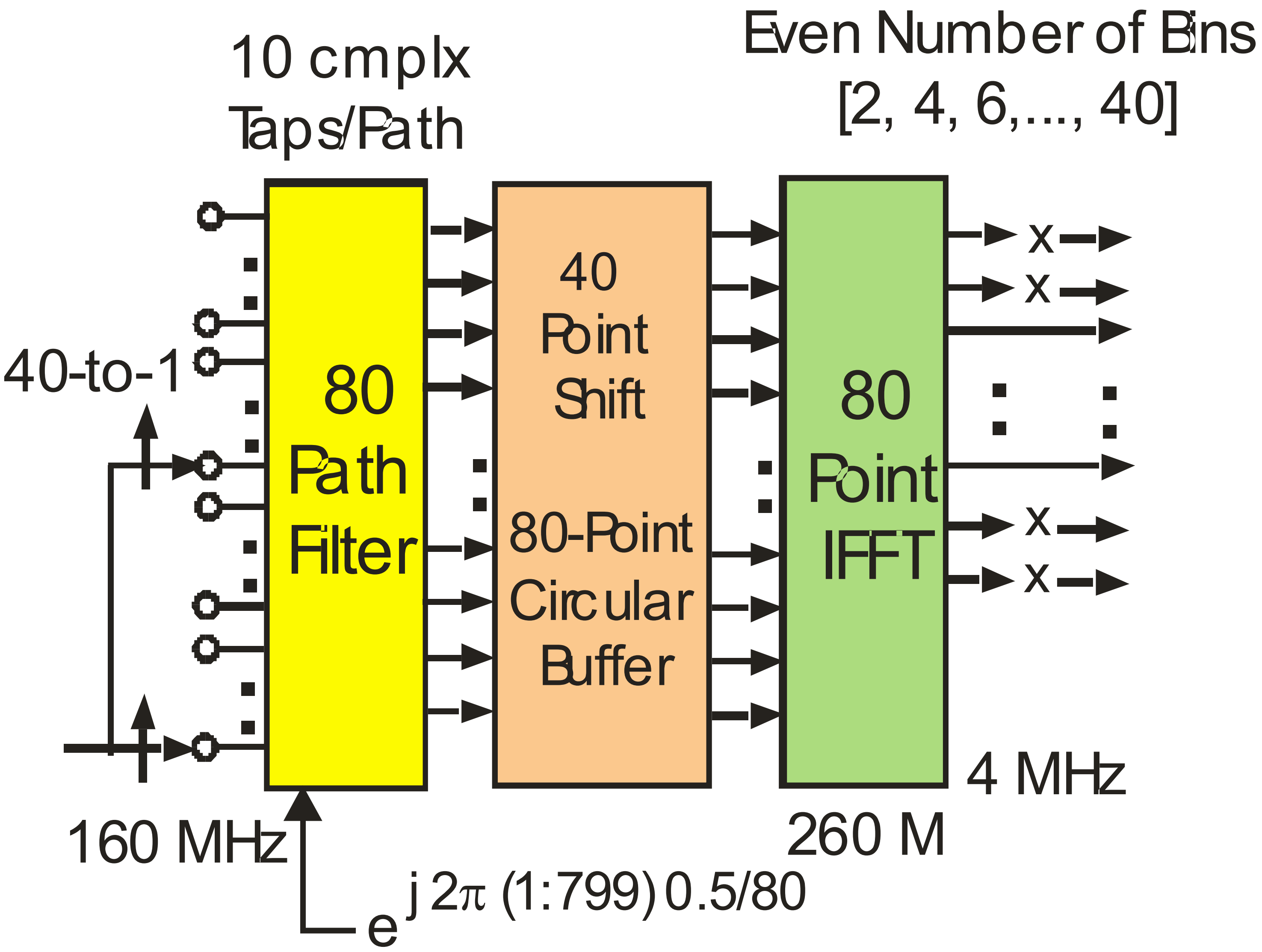


Center Frequencies: Match Roots of Z^N+1, N Roots of -1, $\exp(j\,\pi/N)\exp(j\,2\pi k/N)$



M-Path Analysis Filter Bank with M to 1 Down Sampling Maximally Decimated Perfect Reconstruction Filter Bank Heterodyning Input Time Series Half a Bin Width to Convert Even Indexed to Odd Indexed Bin Centers




10 cmplx
Taps/Path
Even Number of Bins
[2, 4, 6,..., 40]
40-to-1
80
Path
Filter
40
Point
Shift
80-Point
Circular
Buffer
80
Point
IFFT
x
x
x
x
4 MHz
160 MHz
260 M
e j 2π (1:799) 0.5/80

Construct both Even and Odd Interleaved Indexed Spectral Centers from Double Length Polynomial

Even Indexed Bins, $\{\ldots\ -4, -2, 0, +2, +4, \ldots\} \cdot \dfrac{2\pi}{N}$

Located at Roots of $(Z^N - 1)$

Odd Indexed Bins, $\{\ldots\ -5, -3, -1, +1, +3, +5, \ldots\} \cdot \dfrac{2\pi}{N}$

Located at Roots of $(Z^N + 1)$

Interleaved Indicies, $\{\ldots\ -3, -2, -1, 0, +1, +2, +3, \ldots\} \cdot \dfrac{2\pi}{2N}$

Located at Roots of $(Z^N - 1)(Z^{+N} + 1) = (Z^{2N} - 1)$

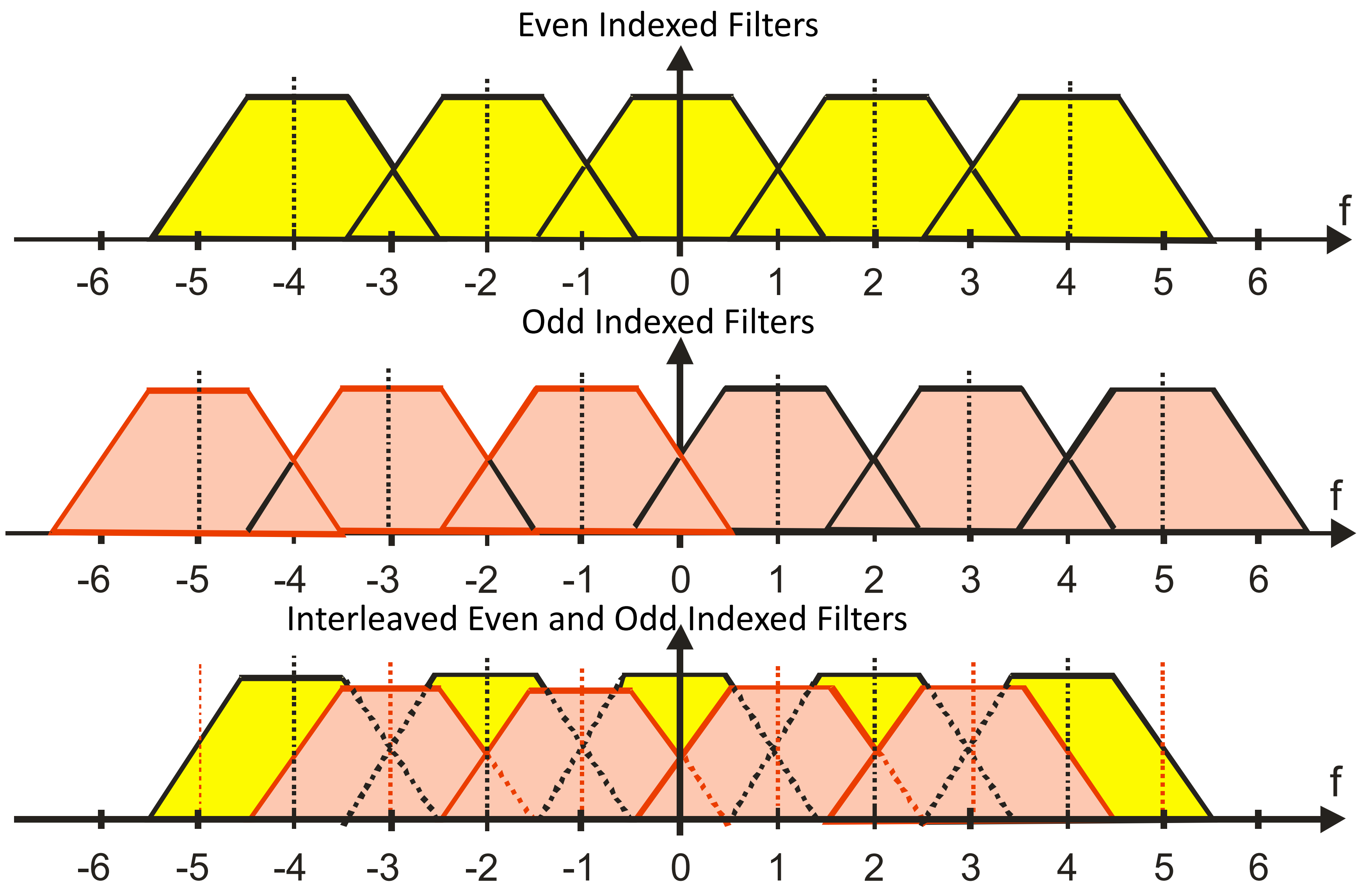
This Option Requires a Double Length Filter and IFFT

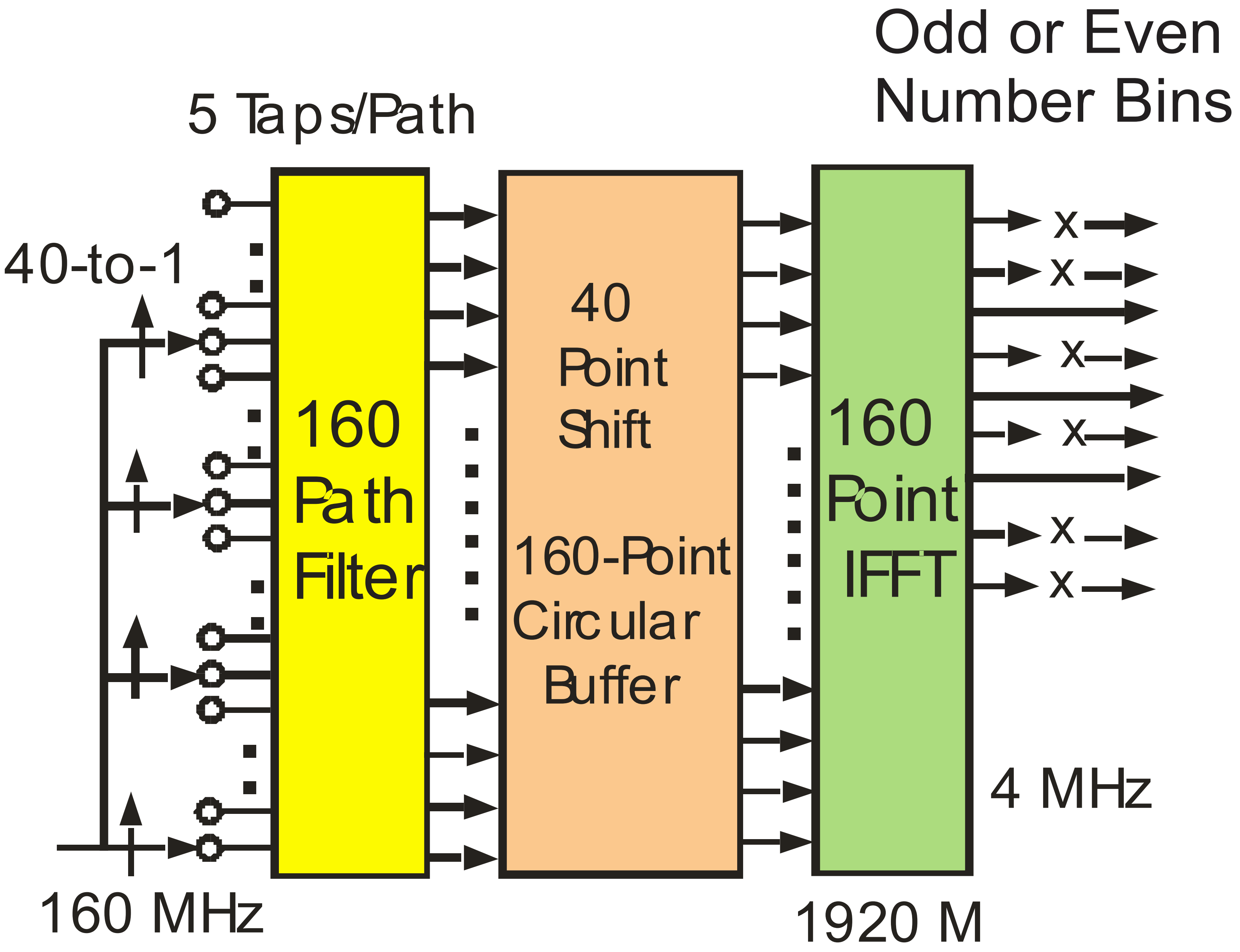
We do away with the Complex Heterodyne of half a

Bin Width of the Input Signal at the cost of a Double Length IFFT

Double Number of Center Frequencies

Center Frequencies: Match Roots of Z^{2N}-1, 2N Roots of 1, exp(j 2π k/(2N))





Odd or Even
Number Bins
5 Taps/Path
40-to-1
160
Path
Filter
40
Point
Shift
160-Point
Circular
Buffer
160
Point
IFFT
x
x
x
x
x
x
4 MHz
160 MHz
1920 M

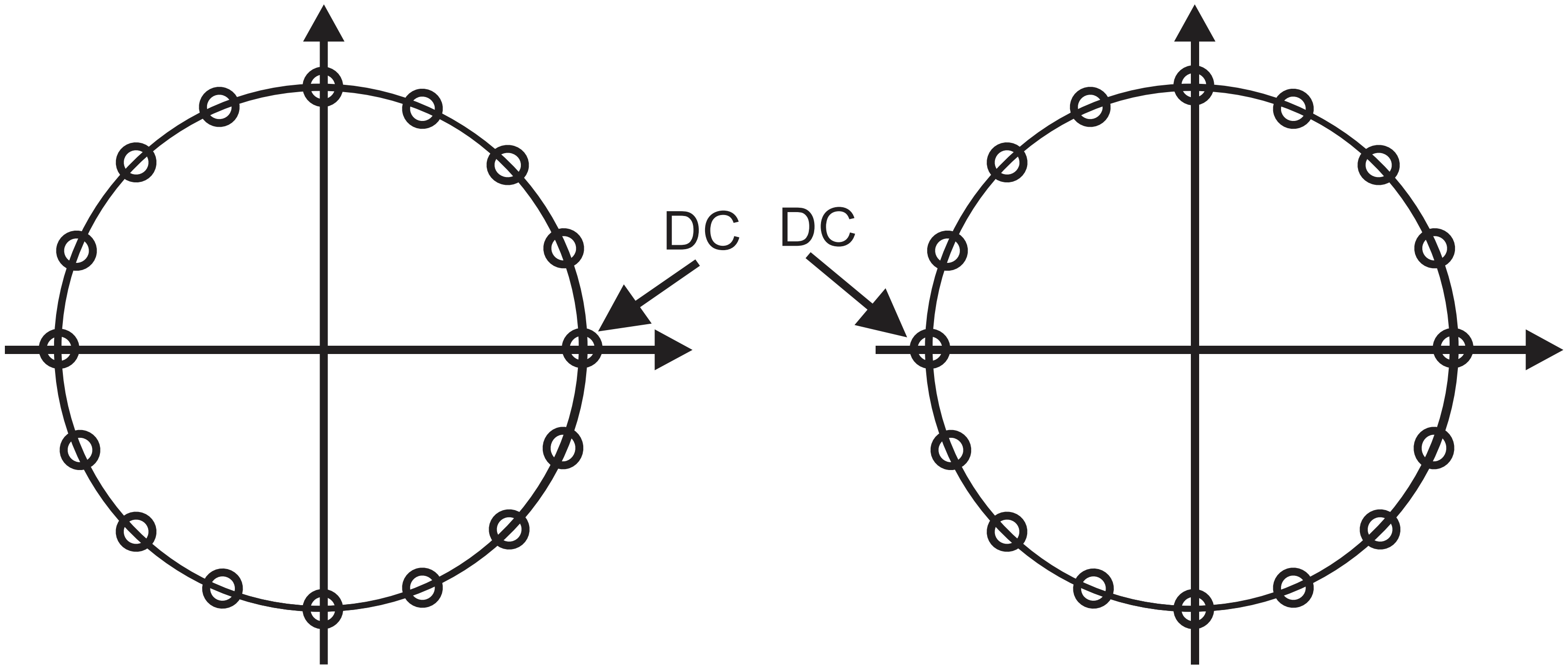
Another Approach

- When N is an Even Number
- There is Root Symmetry about k=0 and k=N/2
- If we Heterodyne DC to the Half Sample Rate

  DC is still on a root of Z^N-1
- DFT Samples Above and Below k=N/2 are the same

  Samples Above and Below k=0, k=+1 and k=-1
- When N is an Odd Number
- The Root Symmetry about k=0 Differs from k=N/2
- There is no Root at k=N/2!
- There is a Root Above and Below N/2 by ±1/2

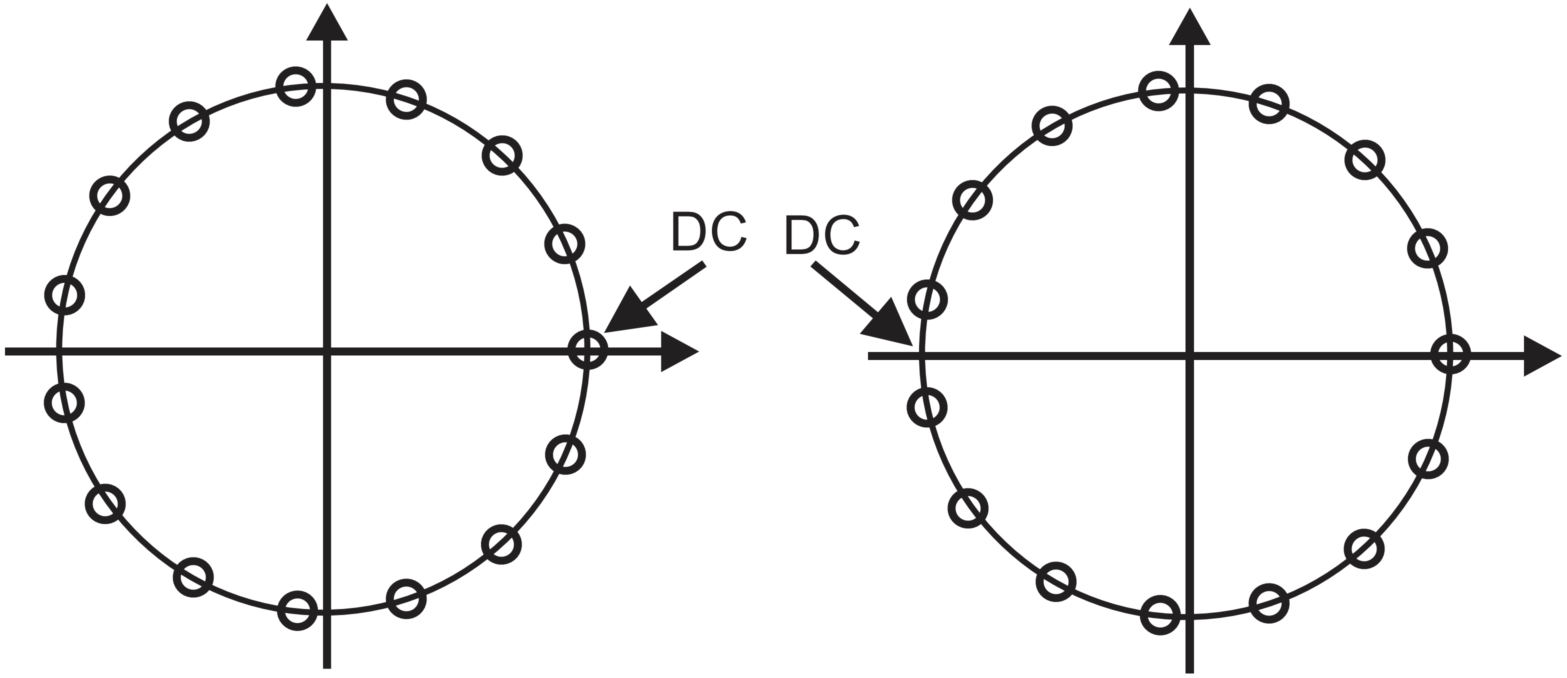
Symmetry of Zeros at DC and at fs/2 of a 16 Point DFT



DC at Index 0
of 16 Point DFT

DC at Index 8
of 16 Point DFT

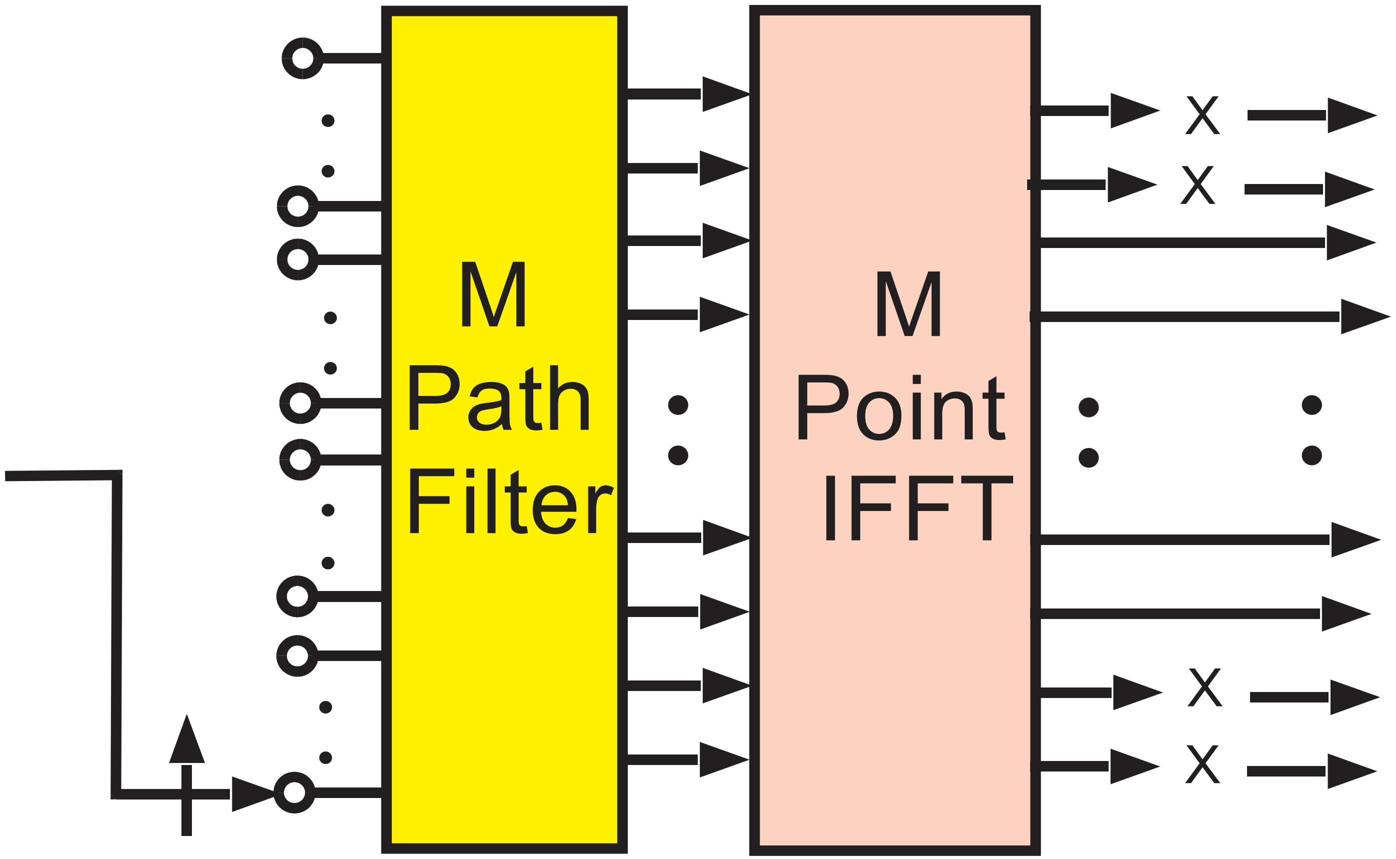
Lack of Symmetry of Zeros at DC and fs/2 of a 15 Point DFT



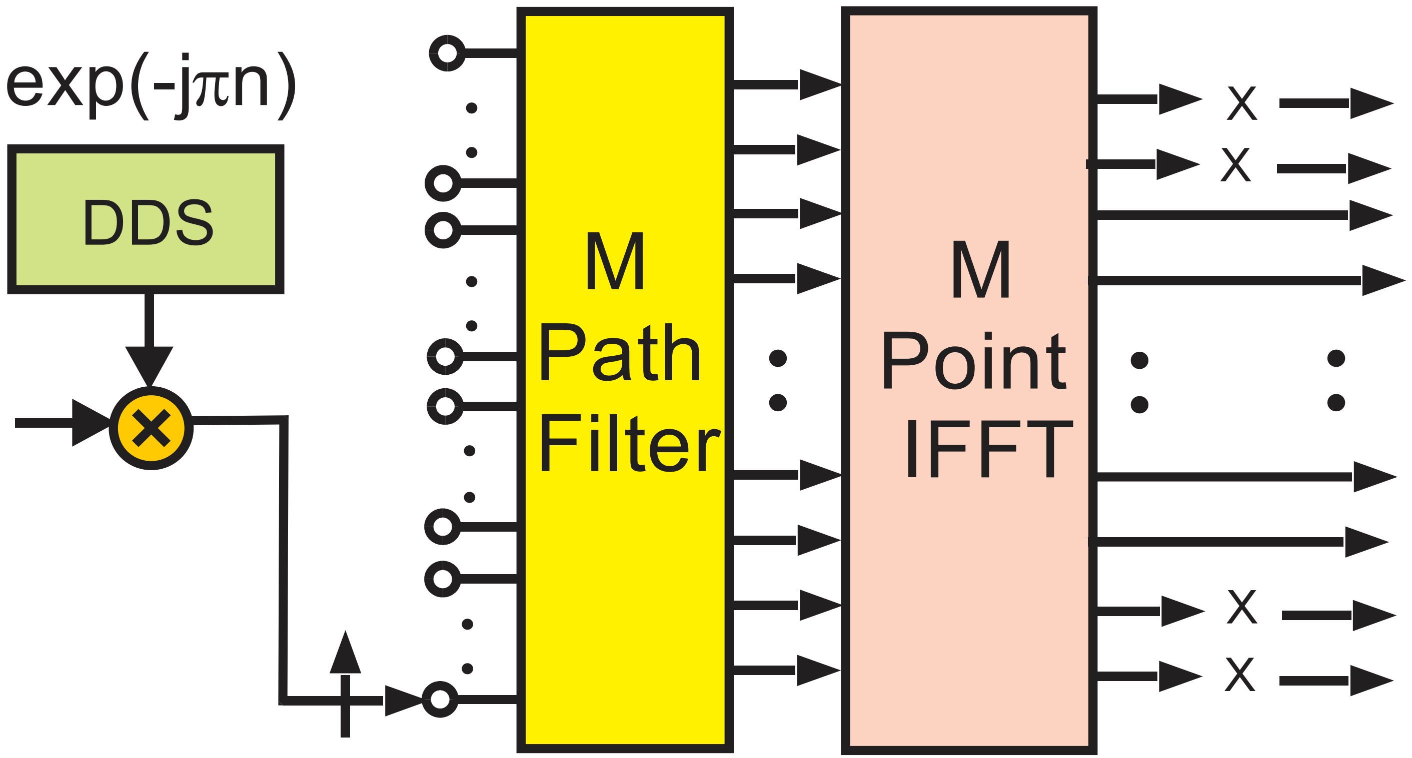
DC at Index 0
of 15 Point DFT

DC at Index 7.5
of 15 Point DFT

M-Path Channelizer For Even Indexed DFT Bin Centers



M-Path Channelizer For Odd Indexed DFT Bin Centers
Don’t Shift Input Spectrum Half a Bin, Shift to fs/2



The Polyphase Filter Sees the Sign Changes of The Input Samples Because the Array has an Odd Number of Samples

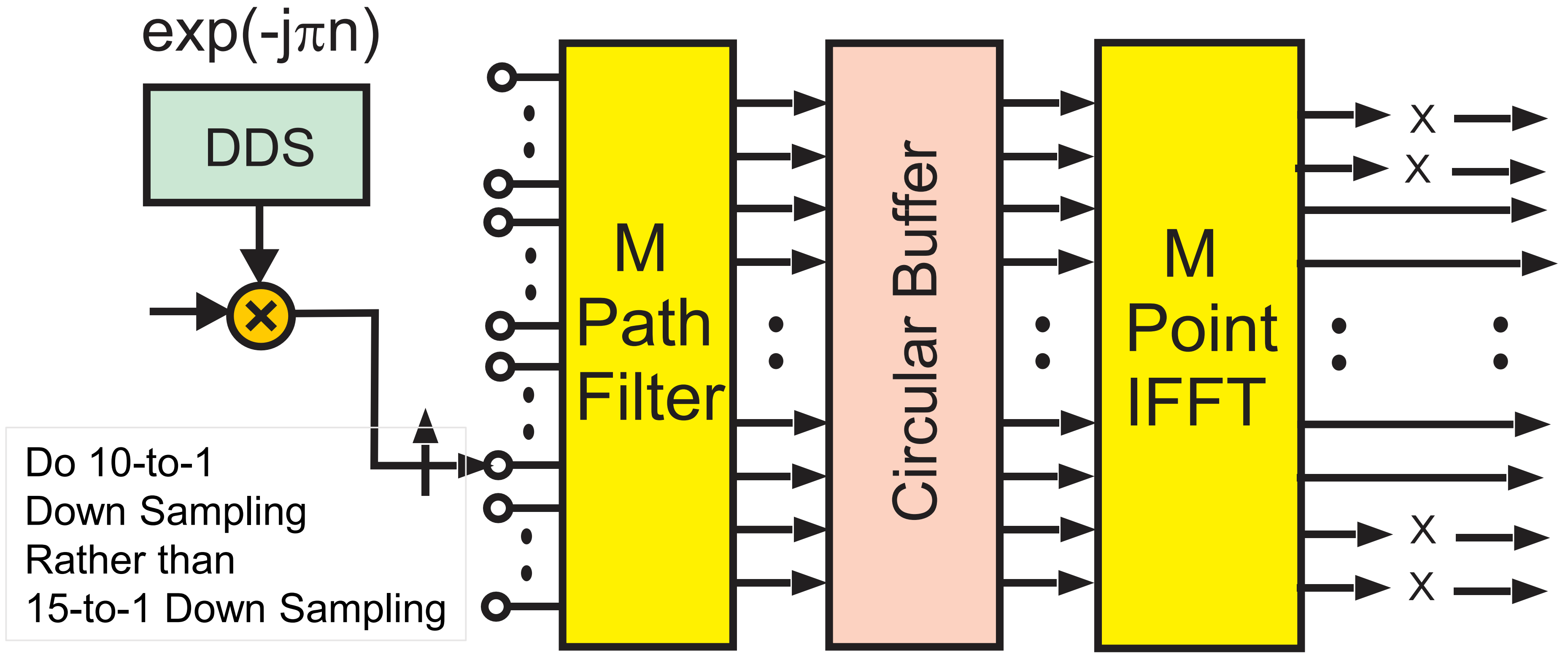
data	sign	data	sign
new		old	
n	+	n-15	-
n-1	-	n-16	+
n-2	+	n-17	-
n-3	-	n-18	+
n-4	+	n-19	-
n-5	-	n-20	+
n-6	+	n-21	-
n-7	-	n-22	+
n-8	+	n-23	-
n-9	-	n-24	+
n-10	+	n-25	-
n-11	-	n-26	+
n-12	+	n-27	-
n-13	-	n-28	+
n-14	+	n-29	-

15 New Inputs with sign

data	sign	data	sign
new		old	
n+15	-	n	+
n+14	+	n-1	-
n+13	-	n-2	+
n+12	+	n-3	-
n+11	-	n-4	+
n+10	+	n-5	-
n+9	-	n-6	+
n+8	+	n-7	-
n+7	-	n-8	+
n+6	+	n-9	-
n+5	-	n-10	+
n+4	+	n-11	-
n+3	-	n-12	+
n+2	+	n-13	-
n+1	-	n-14	+

15 New Inputs with sign

Perform a Non-Maximally decimated Filter with an Even Number of Input Points



The Polyphase Filter No Longer Sees the Sign Changes of The Input Samples Because the Input Array has an Even Number of Samples

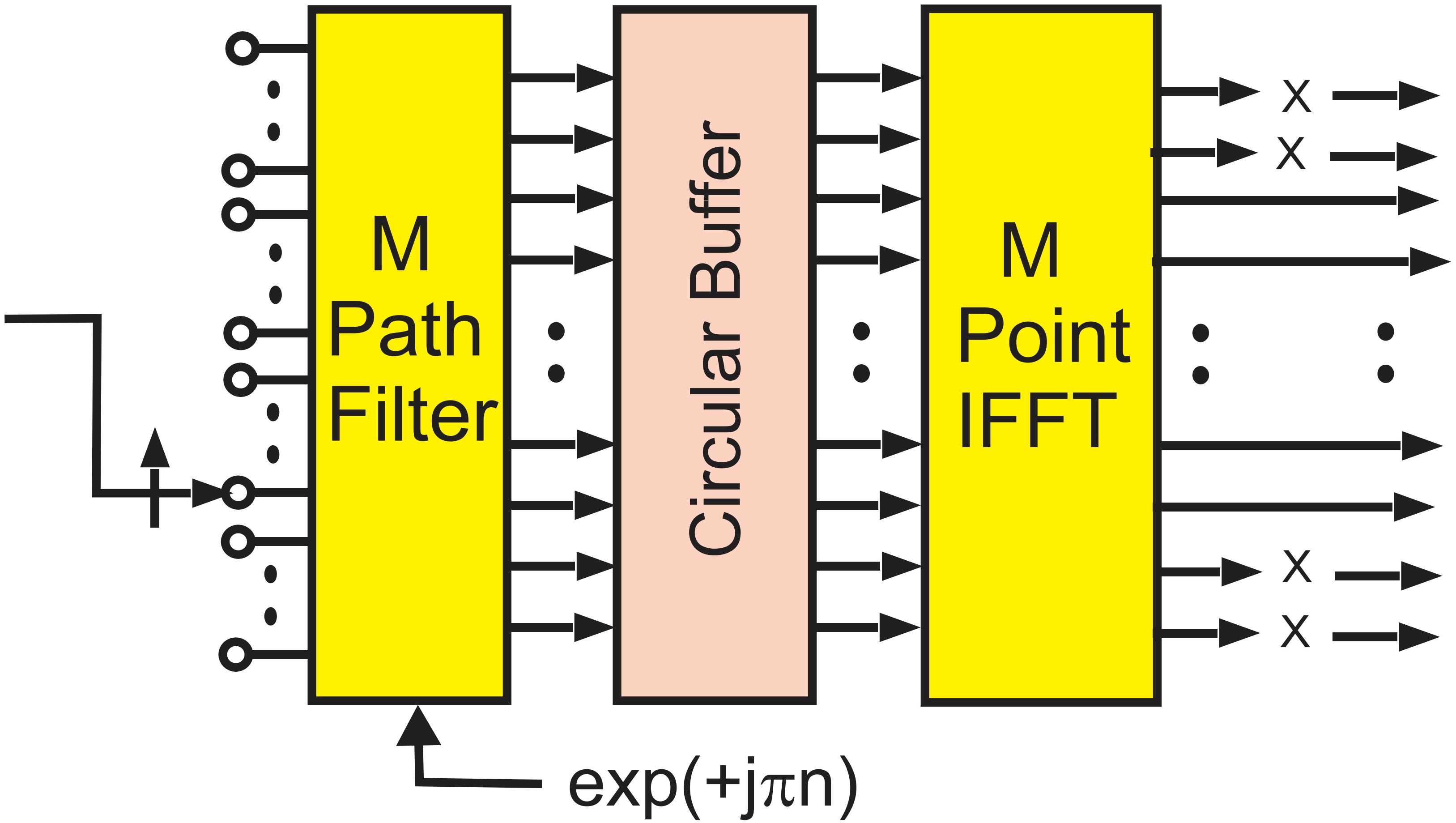
10 New Inputs with sign

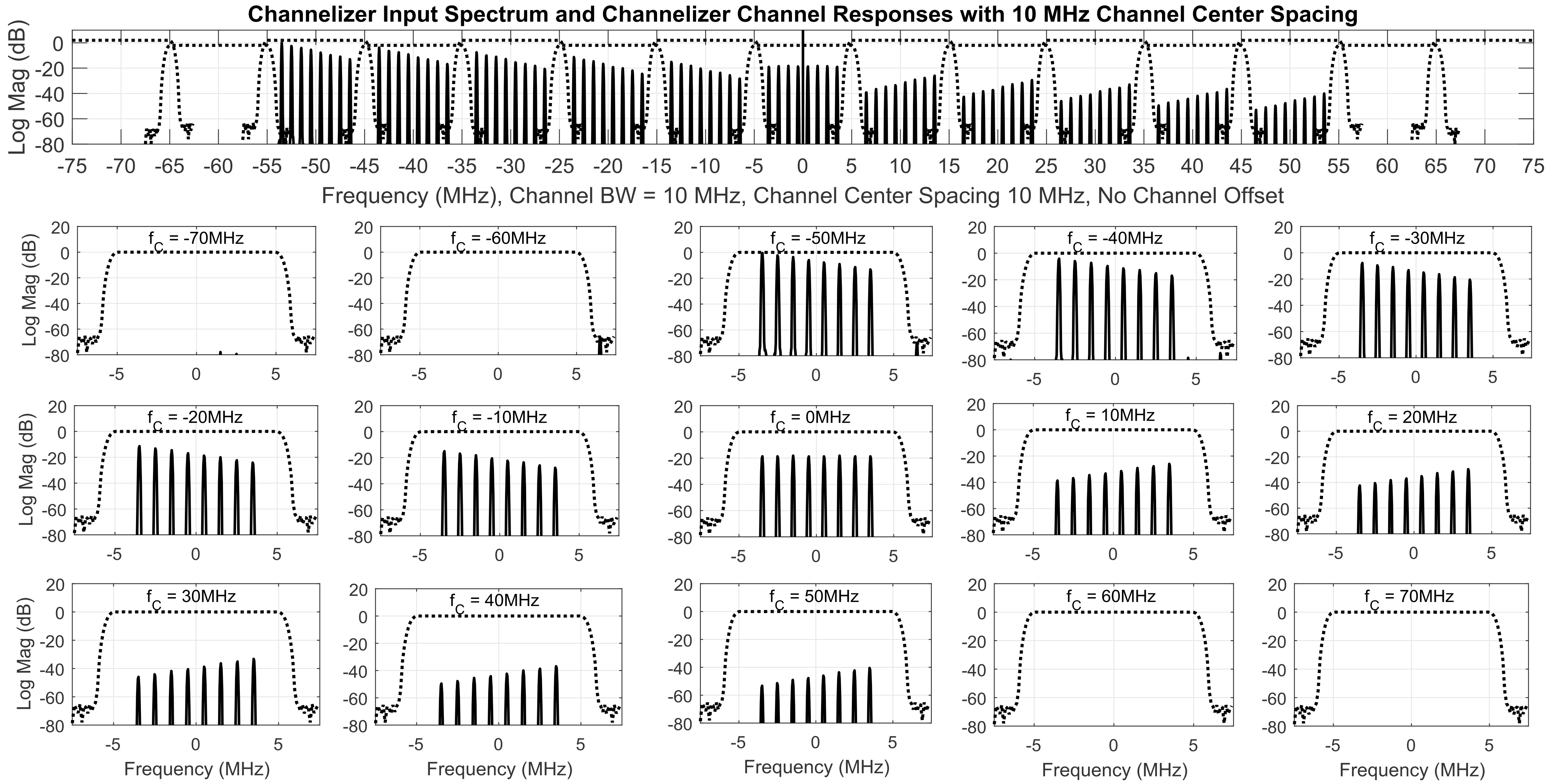
data	sign	data	sign
new		old	
n	+	n-15	-
n-1	-	n-16	+
n-2	+	n-17	-
n-3	-	n-18	+
n-4	+	n-19	-
n-5	-	n-20	+
n-6	+	n-21	-
n-7	-	n-22	+
n-8	+	n-23	-
n-9	-	n-24	+
n-10	+	n-25	-
n-11	-	n-26	+
n-12	+	n-27	-
n-13	-	n-28	+
n-14	+	n-29	-

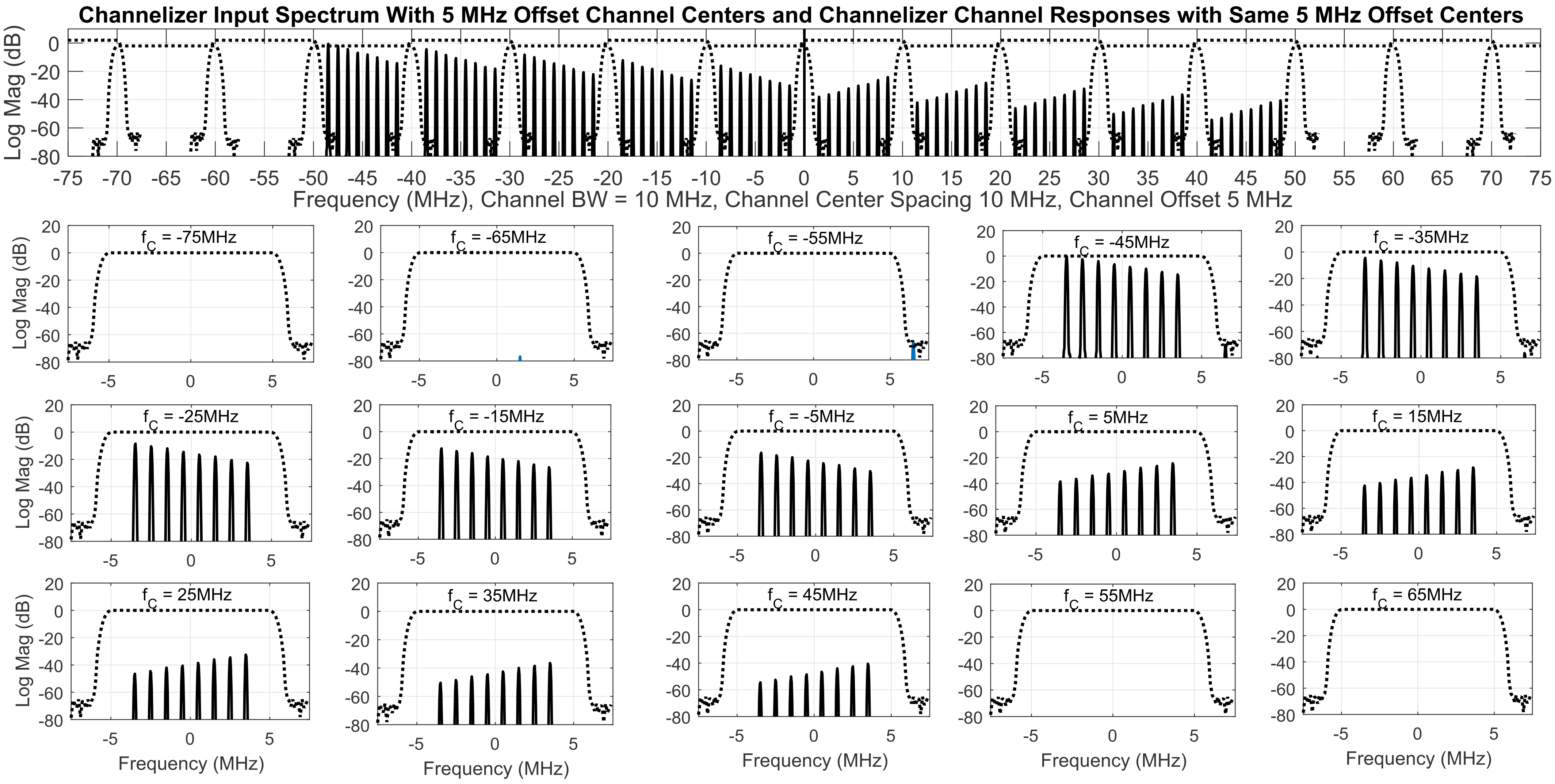
10 New Inputs with sign

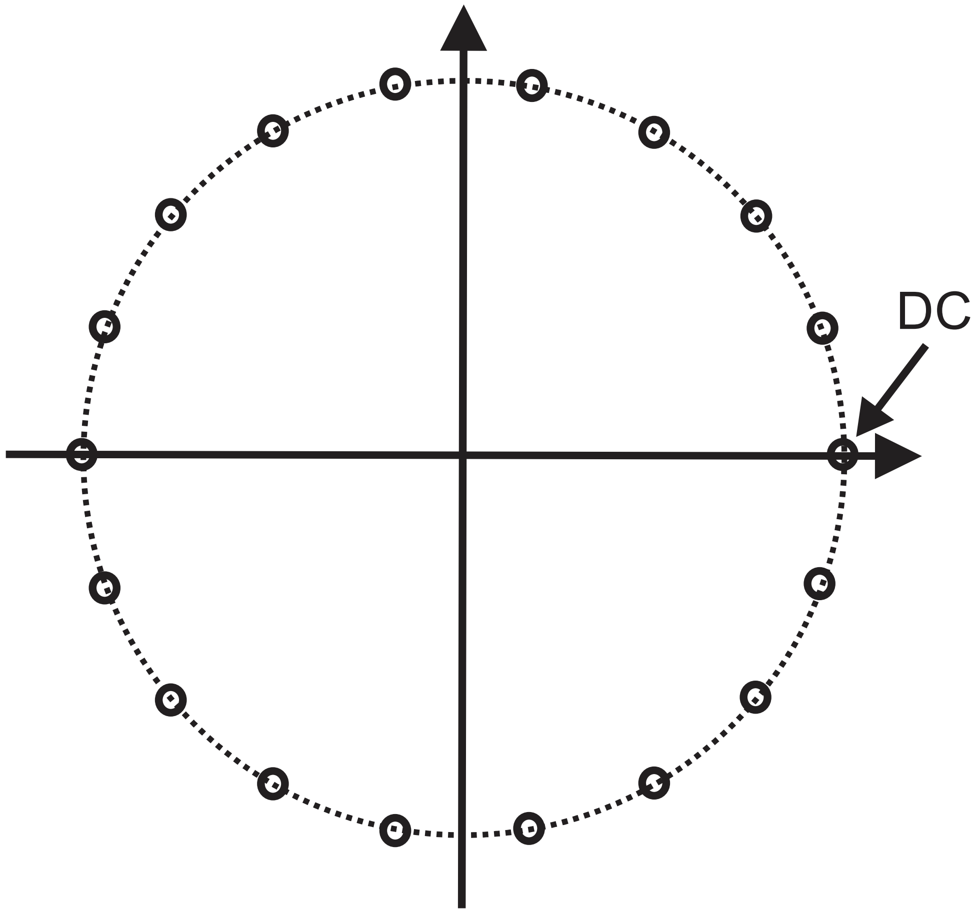
data	sign	data	sign
new		old	
n+10	+	n-5	-
n+9	-	n-6	+
n+8	+	n-7	-
n+7	-	n-8	+
n+6	+	n-9	-
n+5	-	n-10	+
n+4	+	n-11	-
n+3	-	n-12	+
n+2	+	n-13	-
n+1	-	n-14	+
n	+	n-15	-
n-1	-	n-16	+
n-2	+	n-17	-
n-3	-	n-18	+
n-4	+	n-19	-

The Rotators of the Non-Maximally Decimated Filter with an Even Number of Input Points can be Embedded in the Filter Weights




Channelizer Input Spectrum and Channelizer Channel Responses with 10 MHz Channel Center Spacing
Log Mag (dB)
Frequency (MHz), Channel BW = 10 MHz, Channel Center Spacing 10 MHz, No Channel Offset
f_C = -70MHz
f_C = -60MHz
f_C = -50MHz
f_C = -40MHz
f_C = -30MHz
f_C = -20MHz
f_C = -10MHz
f_C = 0MHz
f_C = 10MHz
f_C = 20MHz
f_C = 30MHz
f_C = 40MHz
f_C = 50MHz
f_C = 60MHz
f_C = 70MHz
Frequency (MHz)


Channelizer Input Spectrum With 5 MHz Offset Channel Centers and Channelizer Channel Responses with Same 5 MHz Offset Centers
Log Mag (dB)
Frequency (MHz), Channel BW = 10 MHz, Channel Center Spacing 10 MHz, Channel Offset 5 MHz
f_C = -75MHz
f_C = -65MHz
f_C = -55MHz
f_C = -45MHz
f_C = -35MHz
f_C = -25MHz
f_C = -15MHz
f_C = -5MHz
f_C = 5MHz
f_C = 15MHz
f_C = 25MHz
f_C = 35MHz
f_C = 45MHz
f_C = 55MHz
f_C = 65MHz
Frequency (MHz)



DC at Index 0
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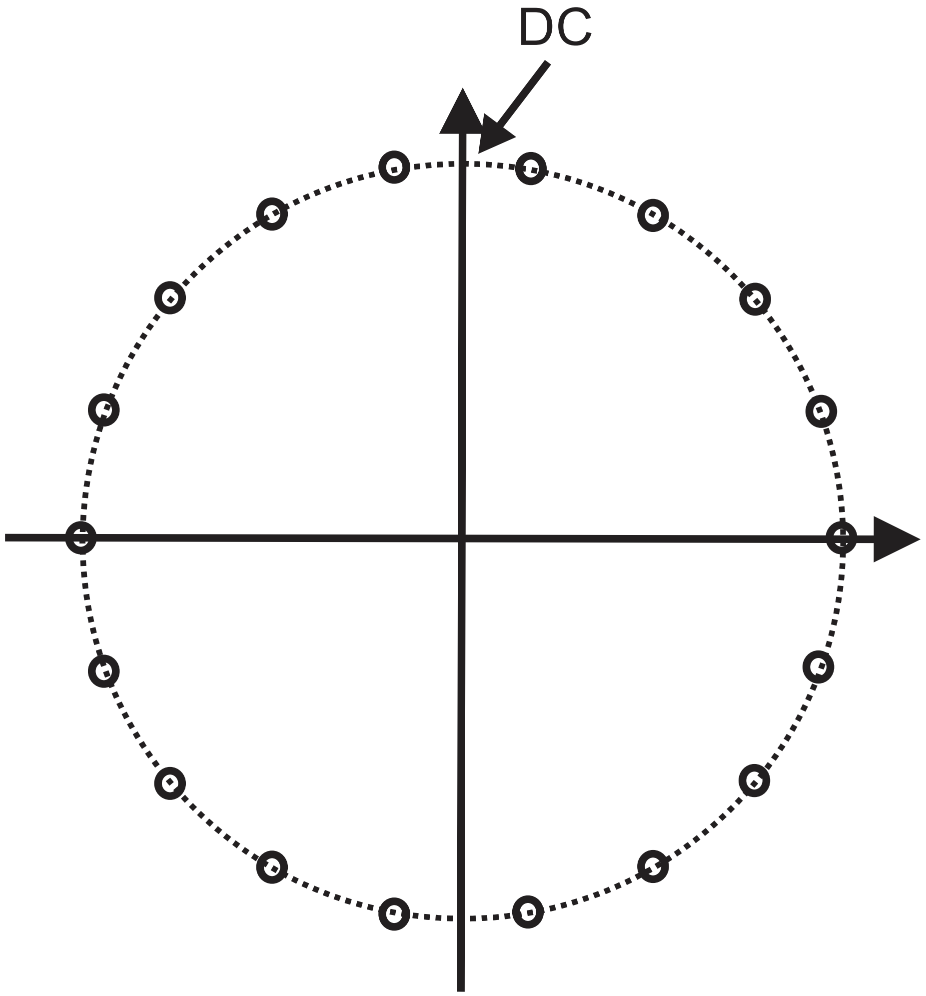

DC at Index 4.5
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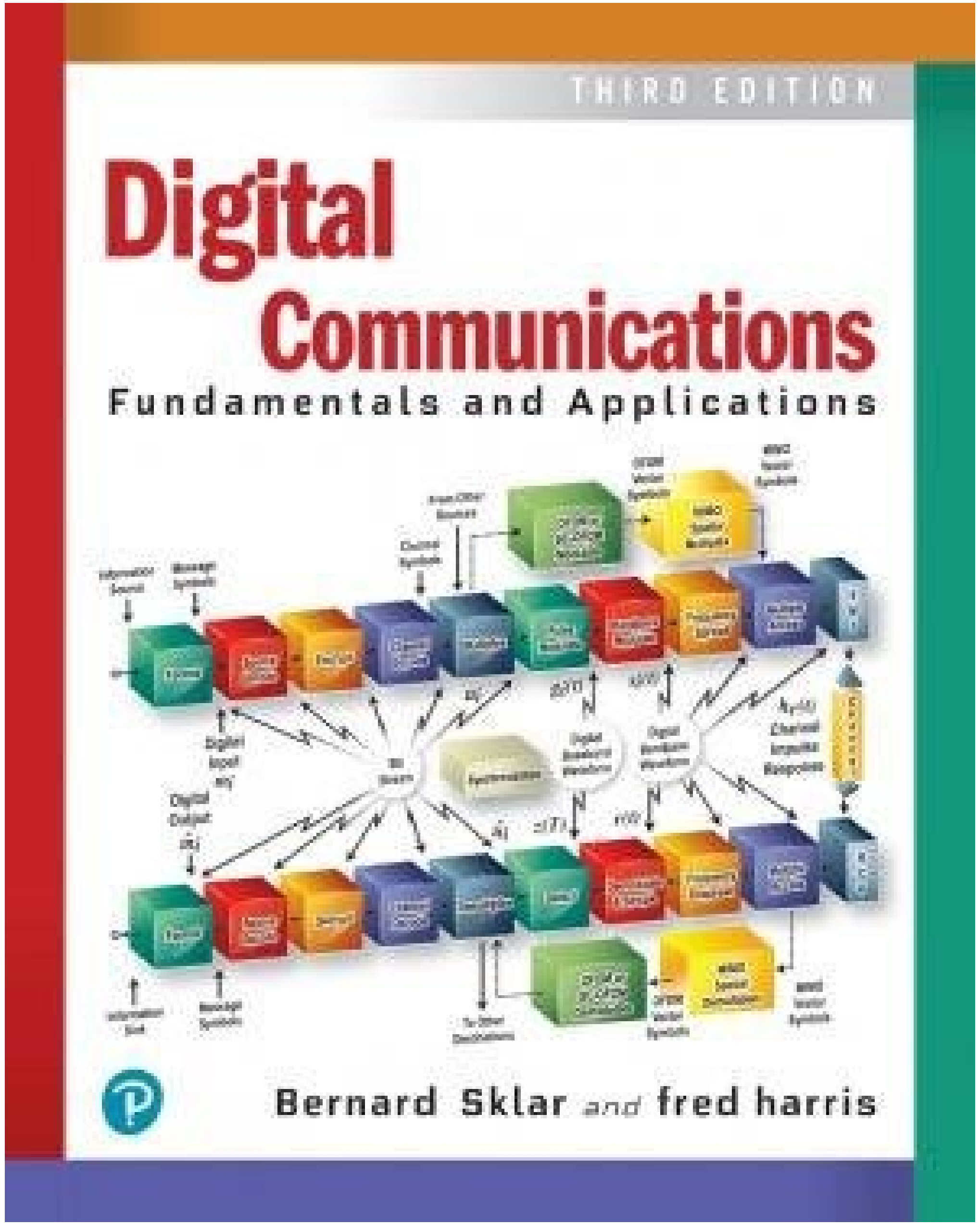
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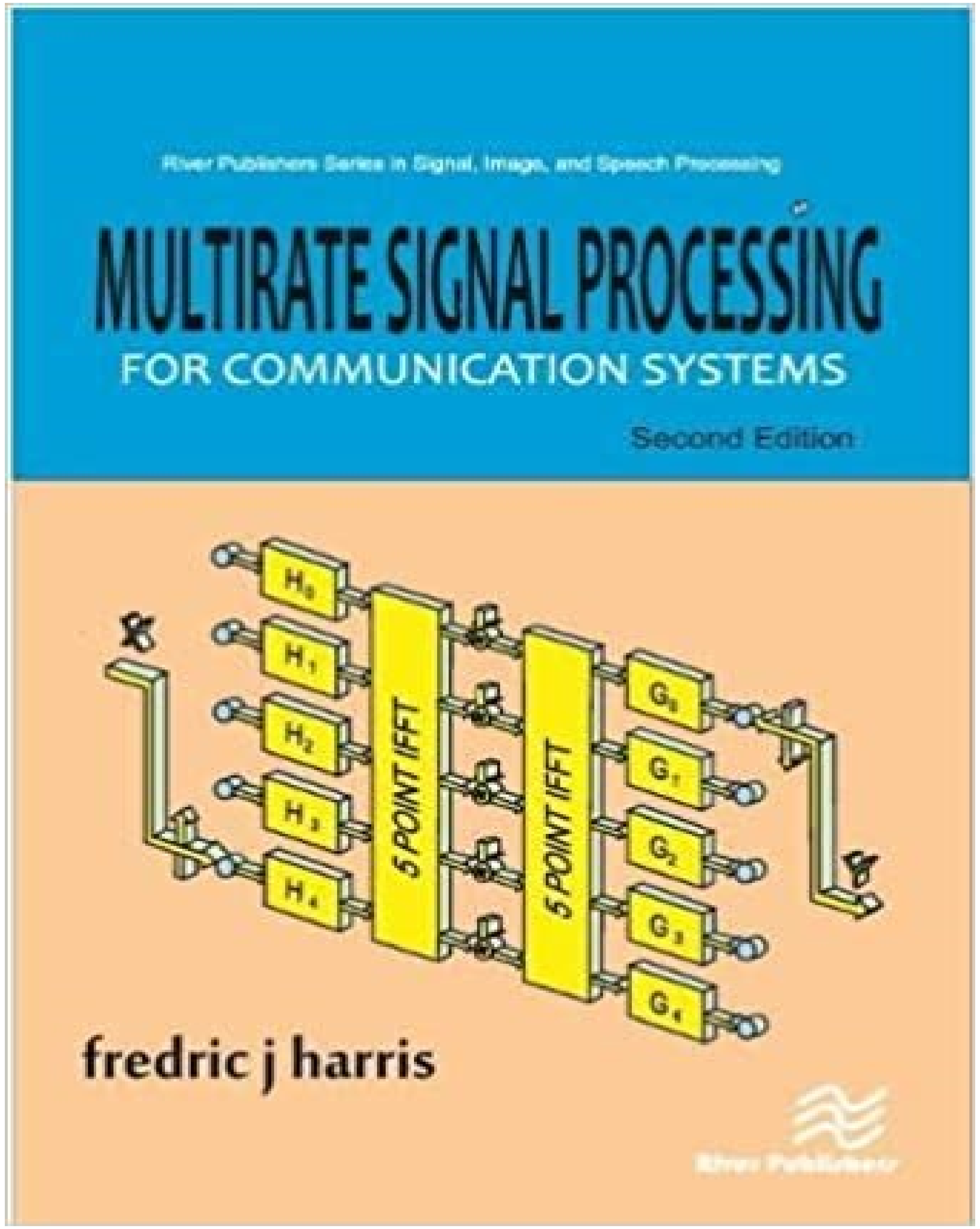
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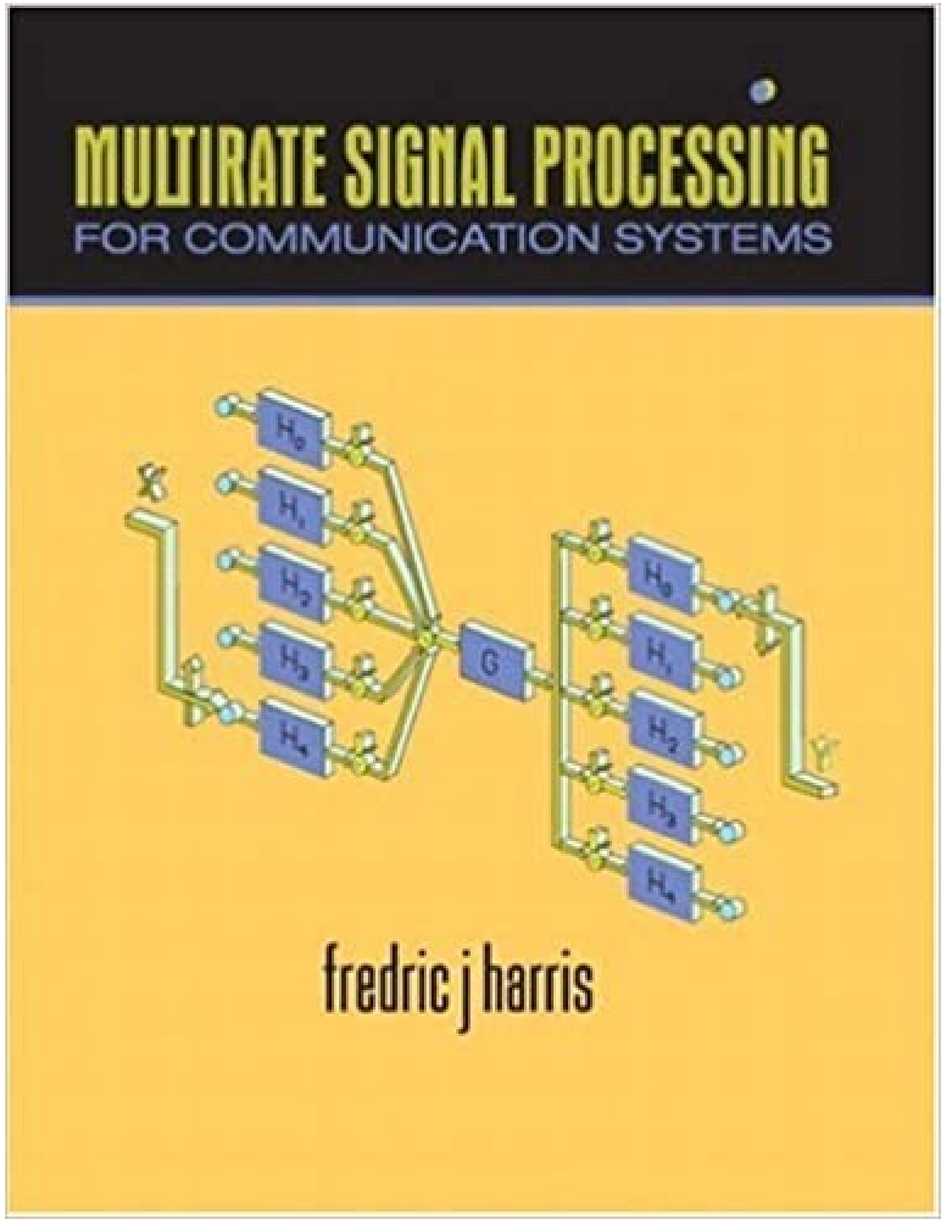
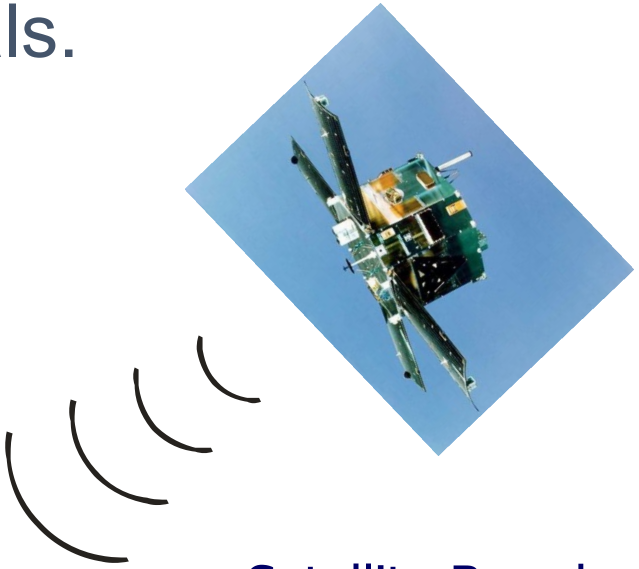


Professor harris: may I be excused?
My brain is full!

Two Quick Examples of Applied Magic with Polyphase Filter Banks

Sitting over Japan is a geosynchronous satellite beaming down 192 stereo MP3 Signals.



Satellite Broadcasts
384 MP3 Channels to
Earth Stations

Demodulate all MP3 Channels
Remodulate as FM Channels

Task: Replace
Legacy Transceiver

What size room is required to
house new DSP based Transceiver?

Equipment Bay: 192-Stereo FM Modulators

Conversation with Client!

- How big a room will we need to house the DSP version of this Transceiver?
- My Answer: I think it will fit on one chip.
- Response:

Don’t be Absurd: Can’t Pack a Room on a Single Chip!

- Results: 48-Analog Devices Blackfin Processors to Demodulate 192 MP3 Stereo Channels.
- 1 Virtex V-4 for 192 Digital Stereo FM Modulators and 256 Channel Channelizer @ 293 kHz Bandwidth per Channel. (60% of Chip)

A Smaller Package

X-Digital Systems

2-U High, Full Rack Width

H 3.5 in, 8.89 cm
W 17.0 in, 43.18 cm
D 9.4 in, 23.88 cm

How to Pack a Room of Analog FM Modulators into a Xilinx FPGA

by Fred Harris
Professor
San Diego State University
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Director of Engineering
Signum Concepts
dragan.vuletic@signumconcepts.com

Wade Lowdermilk
CEO
Signum Concepts
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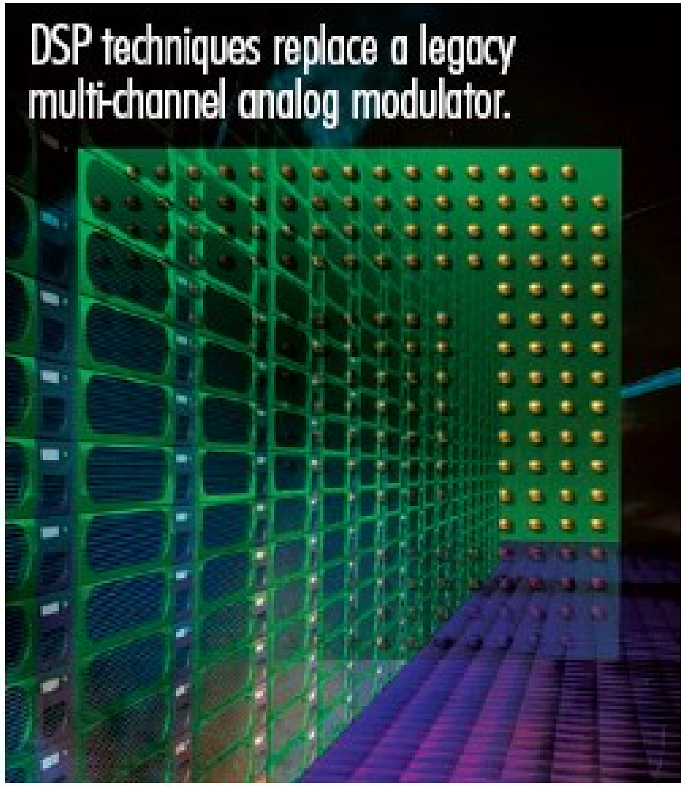

You are likely familiar with the way that digital television is transmitted from satellites as multi-channel MPEG (Motion Picture Experts Group) compressed video to a cable head end where the multiple channels are demodulated. The MPEG streams are decoded and then remodulated as channelized analog NTSC (National Television Standards Committee) or PAL (Phase Alternating Lines) television signals for insertion in a cable distribution plant.

Similarly, high-quality stereo audio is transmitted from a satellite as multi-channel MP3 (MPEG Layer-3) compressed audio to a cable head end where the multiple channels are demodulated. The MP3 streams are decoded and then remodulated as channelized analog FM signals for insertion in a cable distribution plant.

Figure 1 – Equipment bay containing legacy transceiver equipment

如何将一大堆模拟 FM 调制器纳入一个 Xilinx FPGA 中

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Wade Lowdermilk
Signum Concepts
首席执行官
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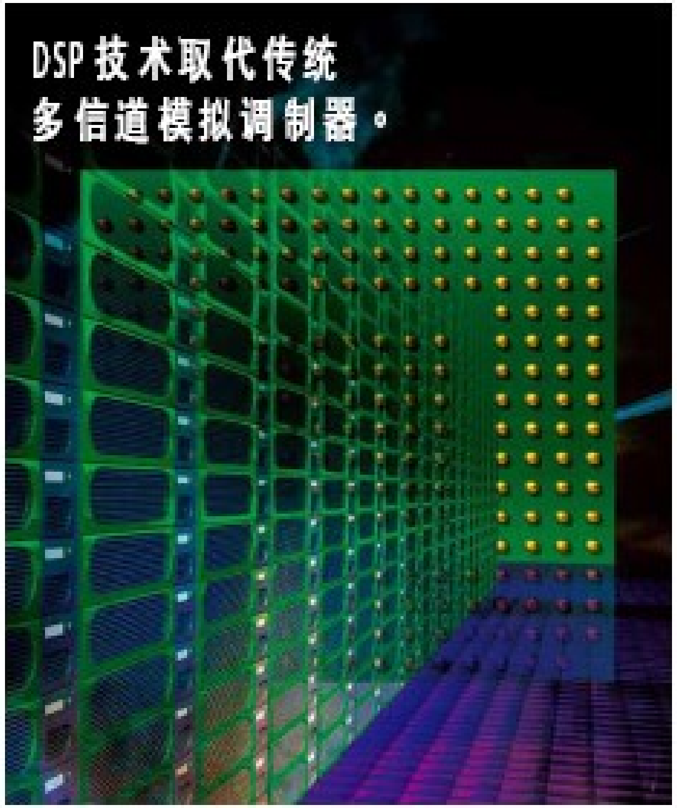


您可能熟悉数字电视的传播方式：信号以多信道 MPEG（移动图像专家组）压缩视频格式从卫星传送到电缆头端器，然后对多个信道进行解调。MPEG 流经解码后被重新调制为信道化模拟 NTSC（国家电视标准委员会）或 PAL（逐行倒相）电视信号并插入到电缆分配设备中。

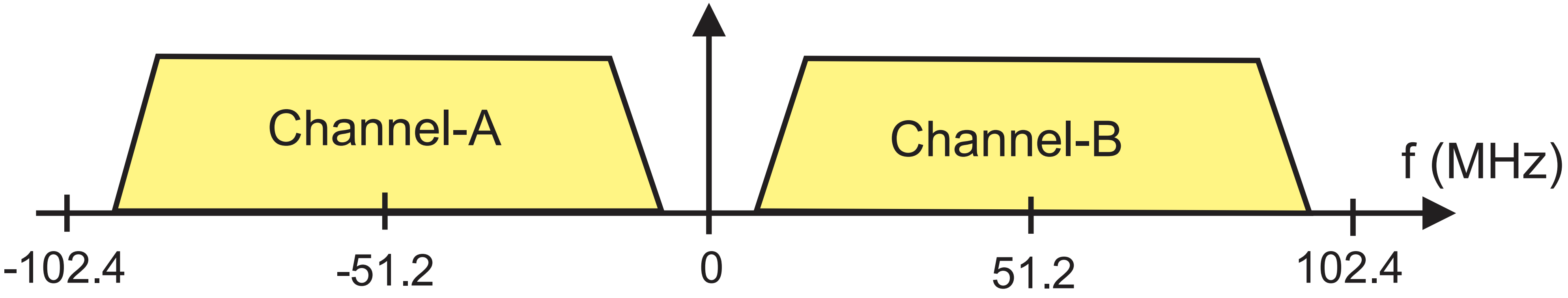
与此相似，高品质的立体声音频信号以多信道 MP3 (MPEG Layer-3) 压缩音频格式从卫星传送到电缆头端器，然后对多个信道进行解调。MP3 流经过解码后被重新调制为信道化模拟 FM 信号并插入到电缆分配设备中。

图 1 - 包含传统收发器设备的设备舱

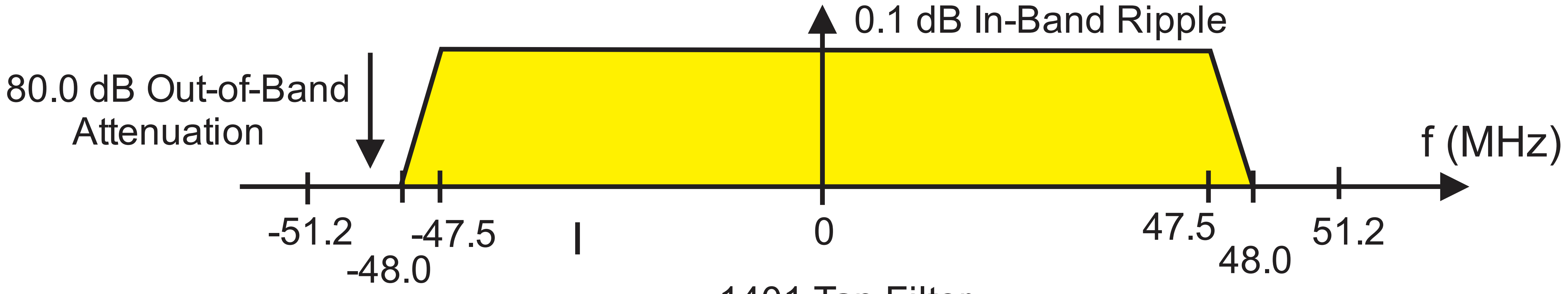
DOCSIS 3.0

Signal Specifications; Complex Input Samples, Independent Upper and Lower 95 MHZ Channels

fs=204.8 MHz



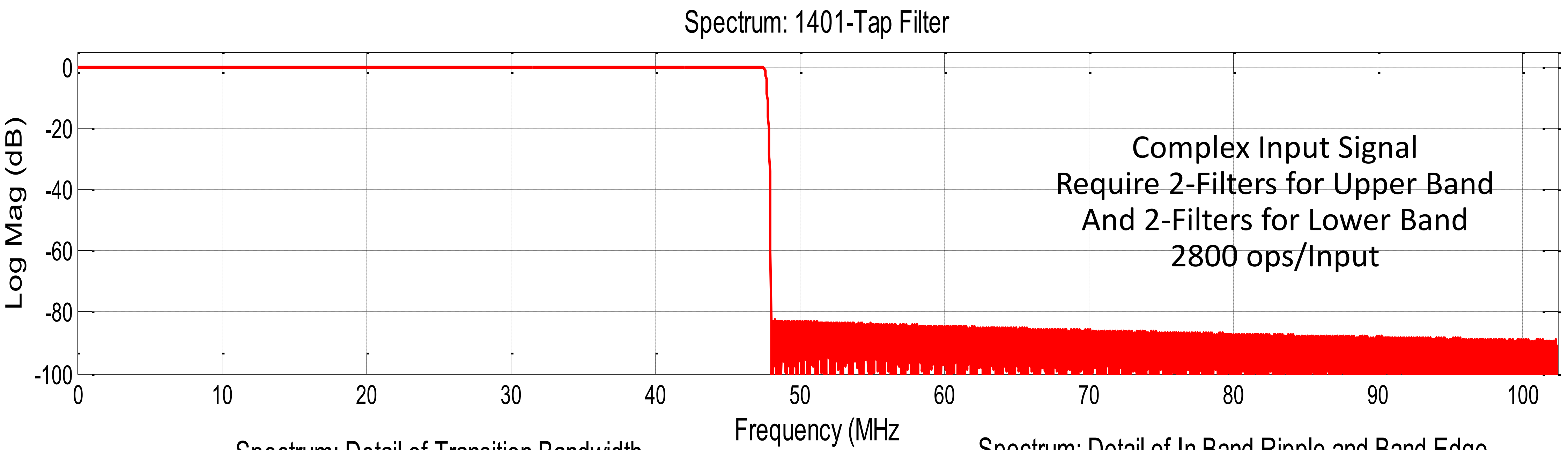
Filter Specifications

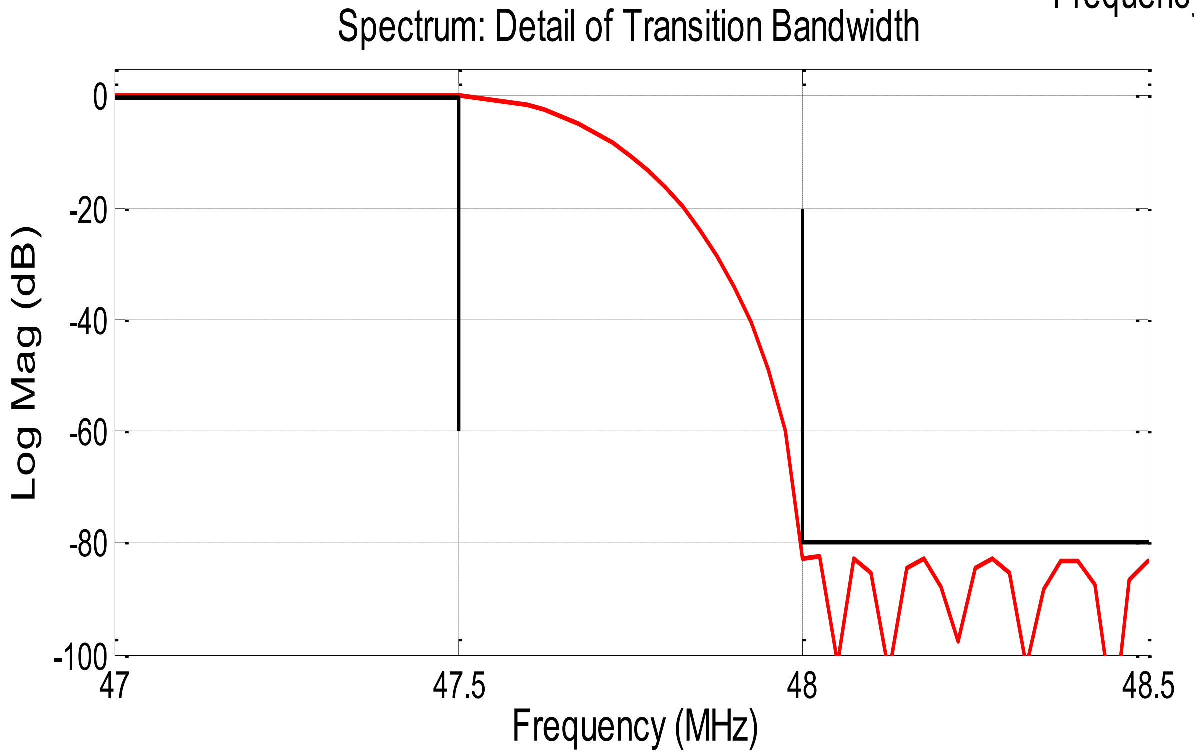


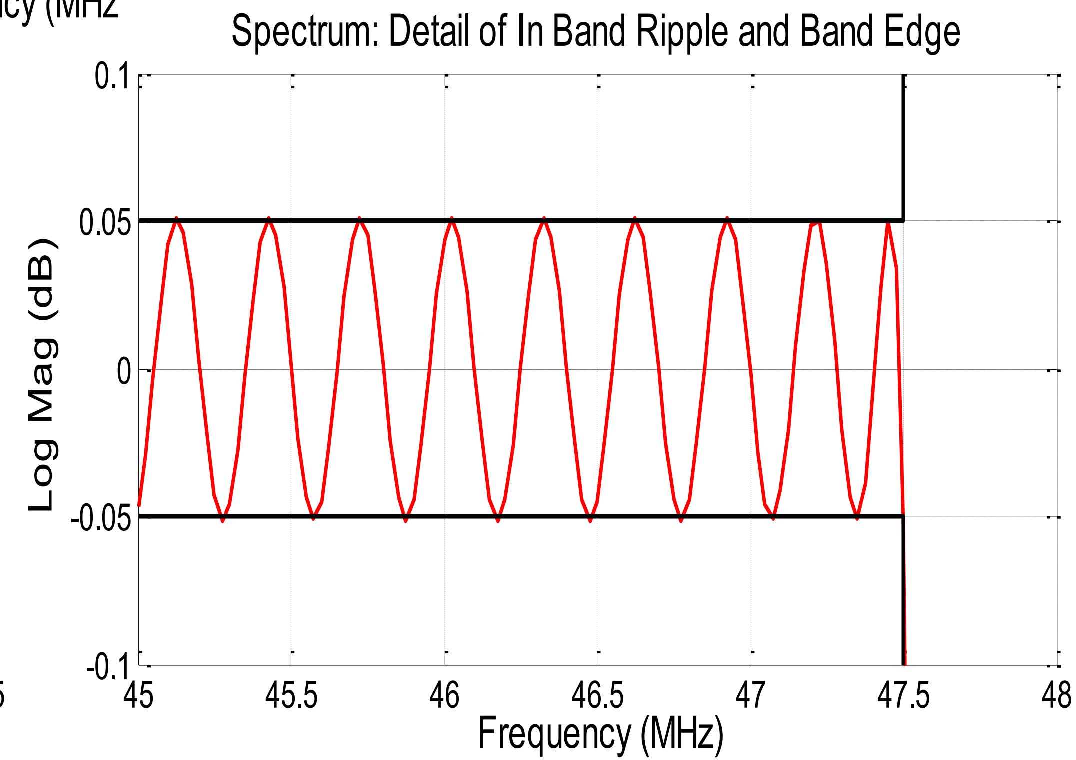
1401 Tap Filter

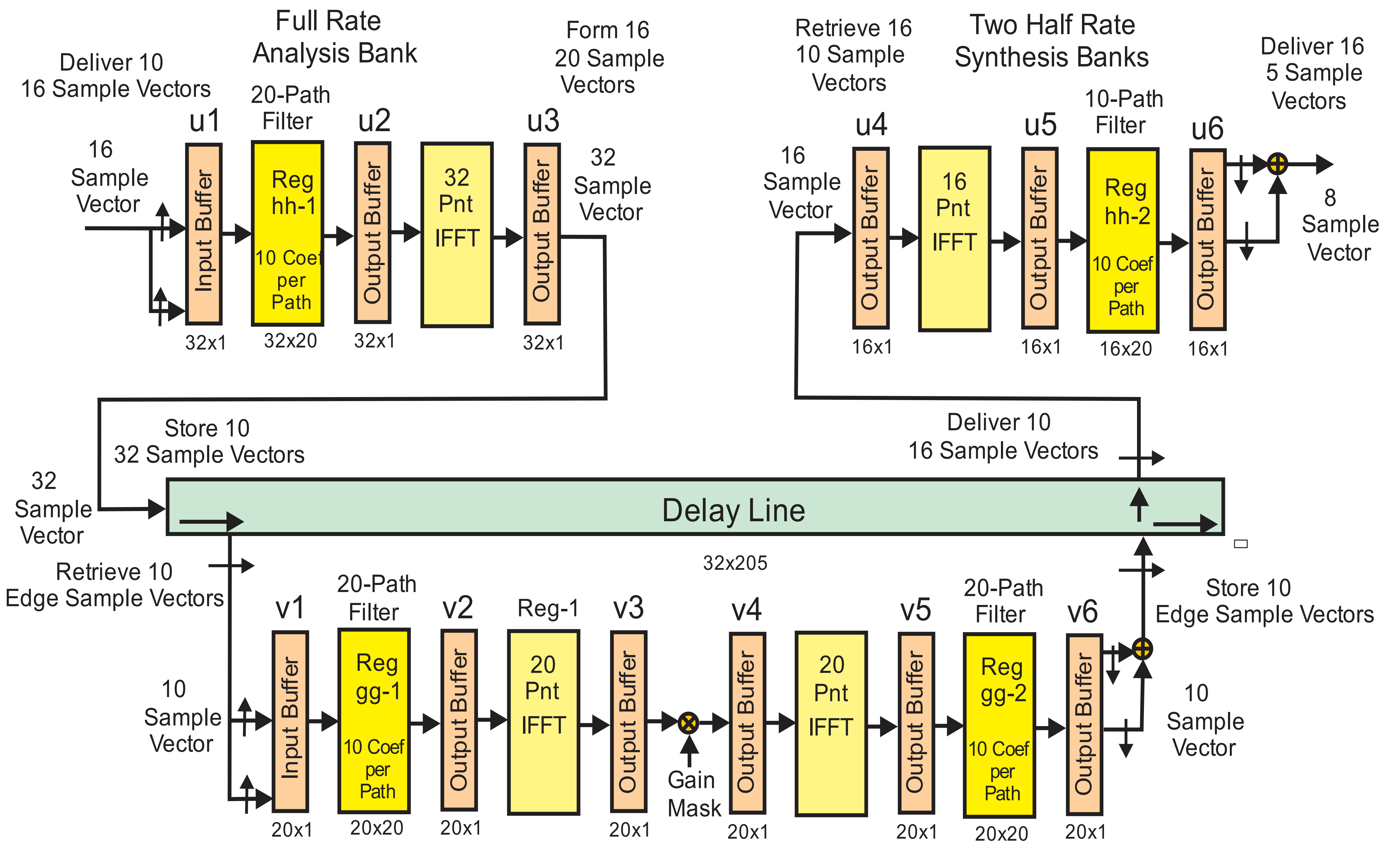
Six Channels, 96 MHz wide Channels, 94 MHz Occupied, 16*102.4 MHz = 1.6384 GHz sample rate

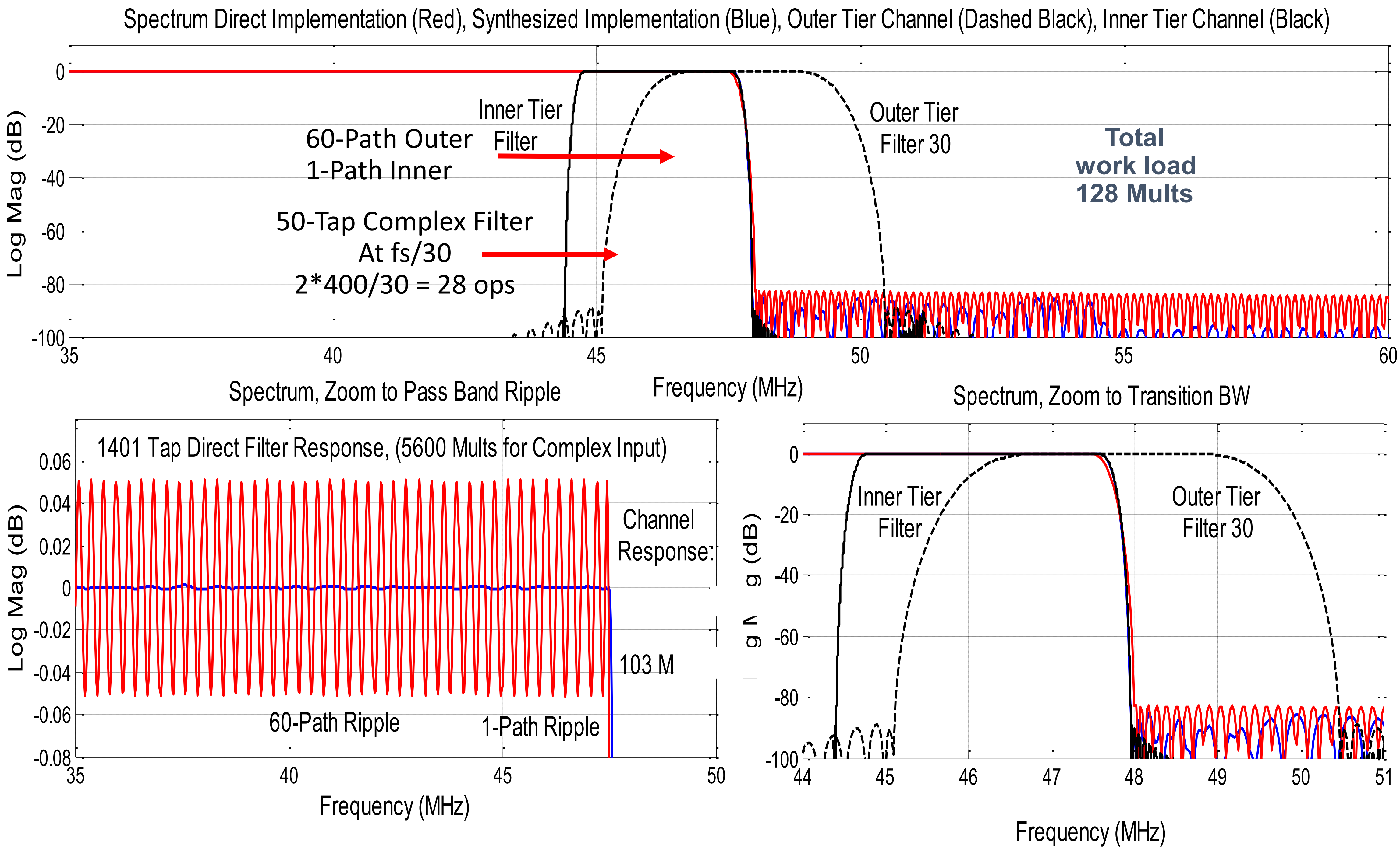
Occupied band 108 to 684 MHz: stopband -80 dB, passband 0.05 dB


Spectrum: 1401-Tap Filter
Complex Input Signal
Require 2-Filters for Upper Band
And 2-Filters for Lower Band
2800 ops/Input
Log Mag (dB)
0
-20
-40
-60
-80
-100
0
10
20
30
40
50
60
70
80
90
100
Frequency (MHz

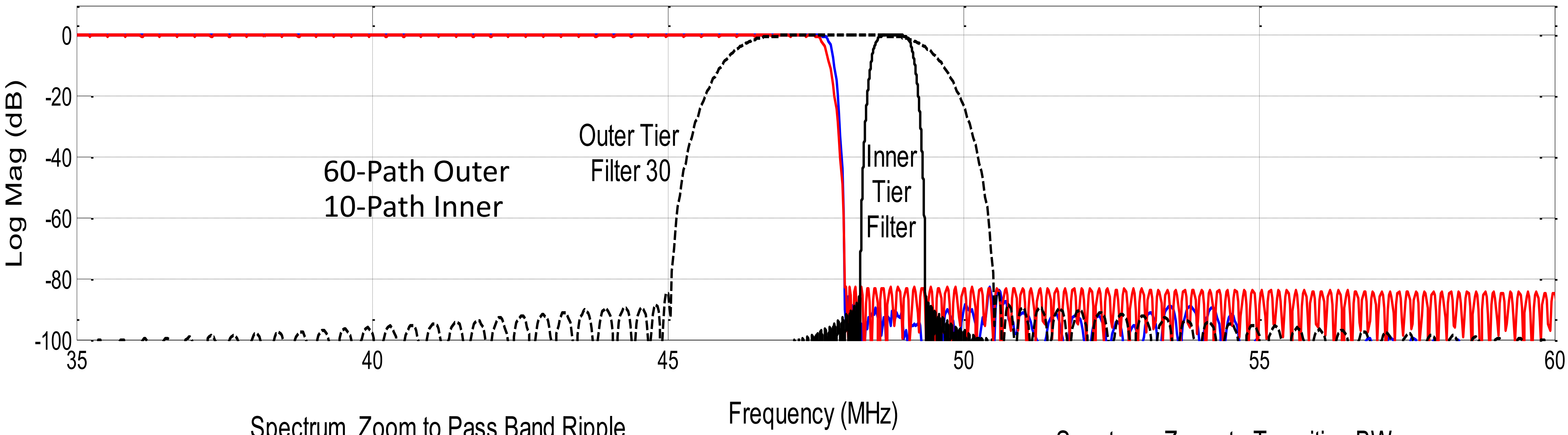

Spectrum: Detail of Transition Bandwidth
Log Mag (dB)
0
-20
-40
-60
-80
-100
47
47.5
48
48.5
Frequency (MHz)


Spectrum: Detail of In Band Ripple and Band Edge
Log Mag (dB)
0.1
0.05
0
-0.05
-0.1
45
45.5
46
46.5
47
47.5
48
Frequency (MHz)

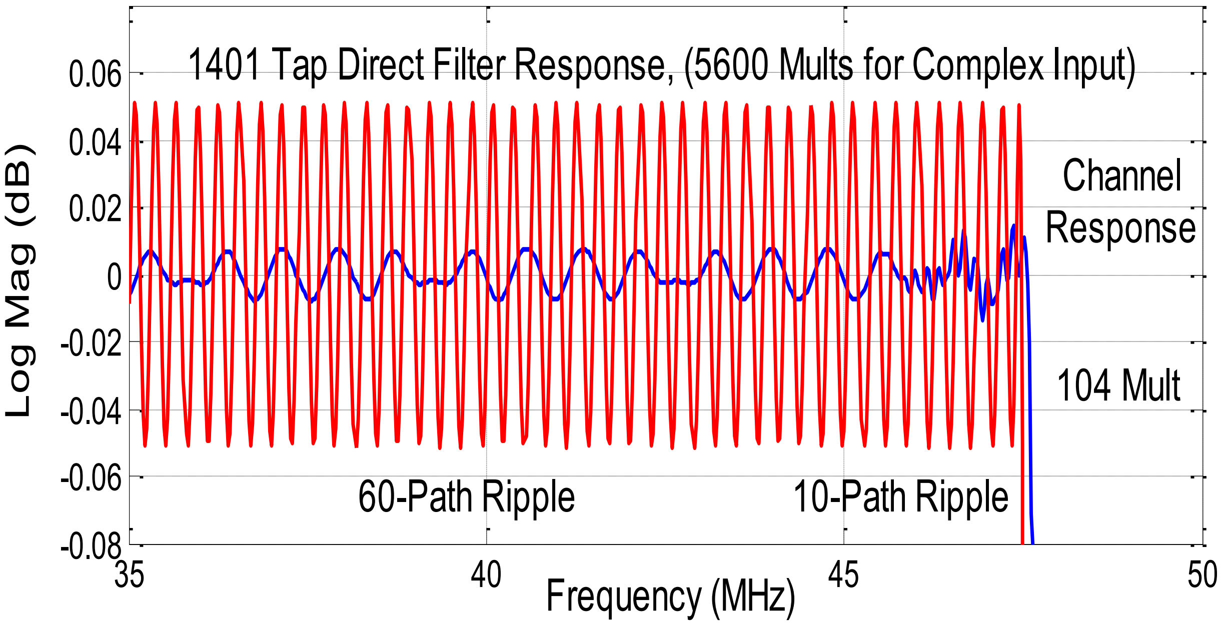

Full Rate
Analysis Bank
Deliver 10
16 Sample Vectors
20-Path
Filter
Form 16
20 Sample
Vectors
u1
u2
u3
16
Sample
Vector
Input Buffer
Reg
hh-1
10 Coef
per
Path
Output Buffer
32
Pnt
IFFT
Output Buffer
32
Sample
Vector
32x1
32x20
32x1
32x1
Retrieve 16
10 Sample
Vectors
Two Half Rate
Synthesis Banks
Deliver 16
5 Sample
Vectors
10-Path
Filter
u4
u5
u6
16
Sample
Vector
Output Buffer
16
Pnt
IFFT
Output Buffer
Reg
hh-2
10 Coef
per
Path
Output Buffer
8
Sample
Vector
16x1
16x1
16x20
16x1
Store 10
32 Sample Vectors
Deliver 10
16 Sample Vectors
32
Sample
Vector
Delay Line
32x205
Retrieve 10
Edge Sample Vectors
Store 10
Edge Sample Vectors
20-Path
Filter
Reg-1
20-Path
Filter
v1
v2
v3
v4
v5
v6
10
Sample
Vector
Input Buffer
Reg
gg-1
10 Coef
per
Path
Output Buffer
20
Pnt
IFFT
Output Buffer
Gain
Mask
Output Buffer
20
Pnt
IFFT
Output Buffer
Reg
gg-2
10 Coef
per
Path
Output Buffer
10
Sample
Vector
20x1
20x20
20x1
20x1
20x1
20x1
20x20
20x1


Spectrum Direct Implementation (Red), Synthesized Implementation (Blue), Outer Tier Channel (Dashed Black), Inner Tier Channel (Black)
Log Mag (dB)
Inner Tier
Filter
60-Path Outer
1-Path Inner
50-Tap Complex Filter
At fs/30
2*400/30 = 28 ops
Outer Tier
Filter 30
Total
work load
128 Mults
0
-20
-40
-60
-80
-100
35
40
45
50
55
60
Frequency (MHz)
Spectrum, Zoom to Pass Band Ripple
1401 Tap Direct Filter Response, (5600 Mults for Complex Input)
Channel
Response:
103 M
60-Path Ripple
1-Path Ripple
0.06
0.04
0.02
0
-0.02
-0.04
-0.06
-0.08
35
40
45
50
Frequency (MHz)
Spectrum, Zoom to Transition BW
Inner Tier
Filter
Outer Tier
Filter 30
0
-20
-40
-60
-80
-100
44
45
46
47
48
49
50
51
Frequency (MHz)

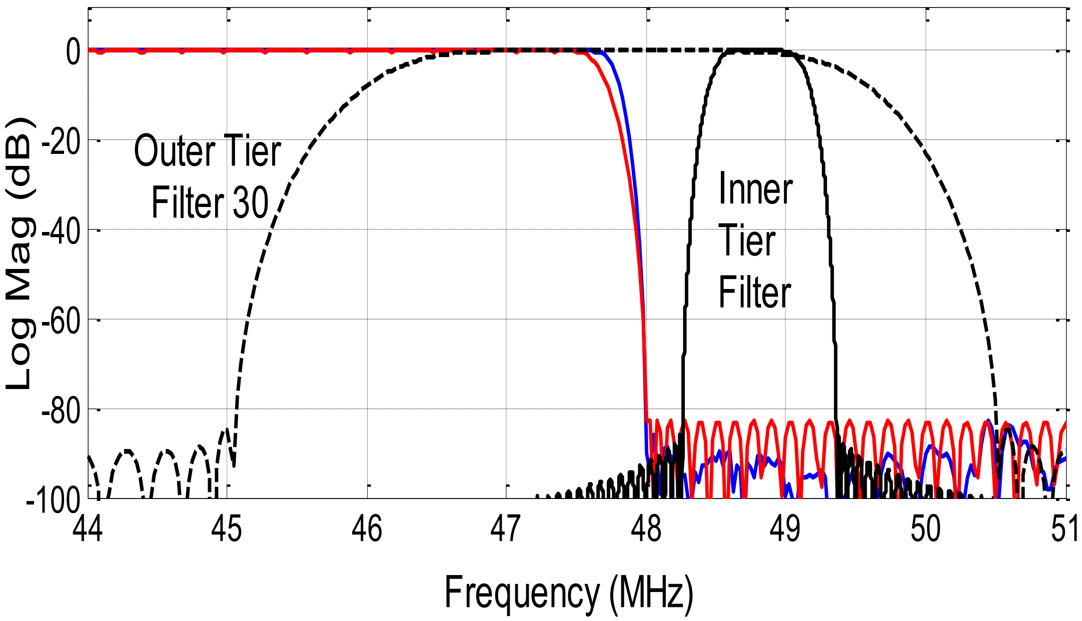
Spectrum Direct Implementation (Red), Synthesized Implementation (Blue), Outer Tier Channel (Dashed Black), Inner Tier Channel (Black)



Spectrum, Zoom to Pass Band Ripple

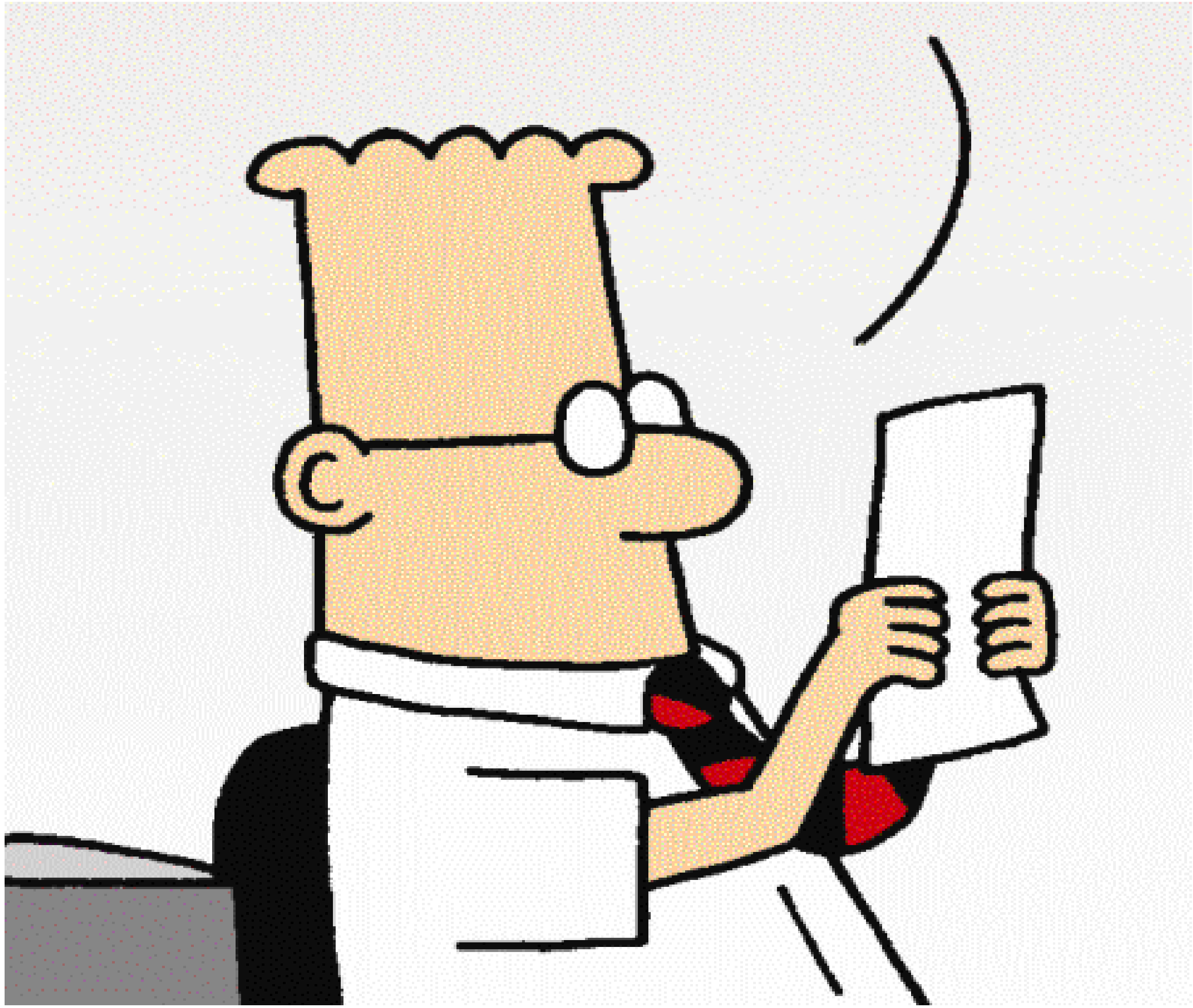


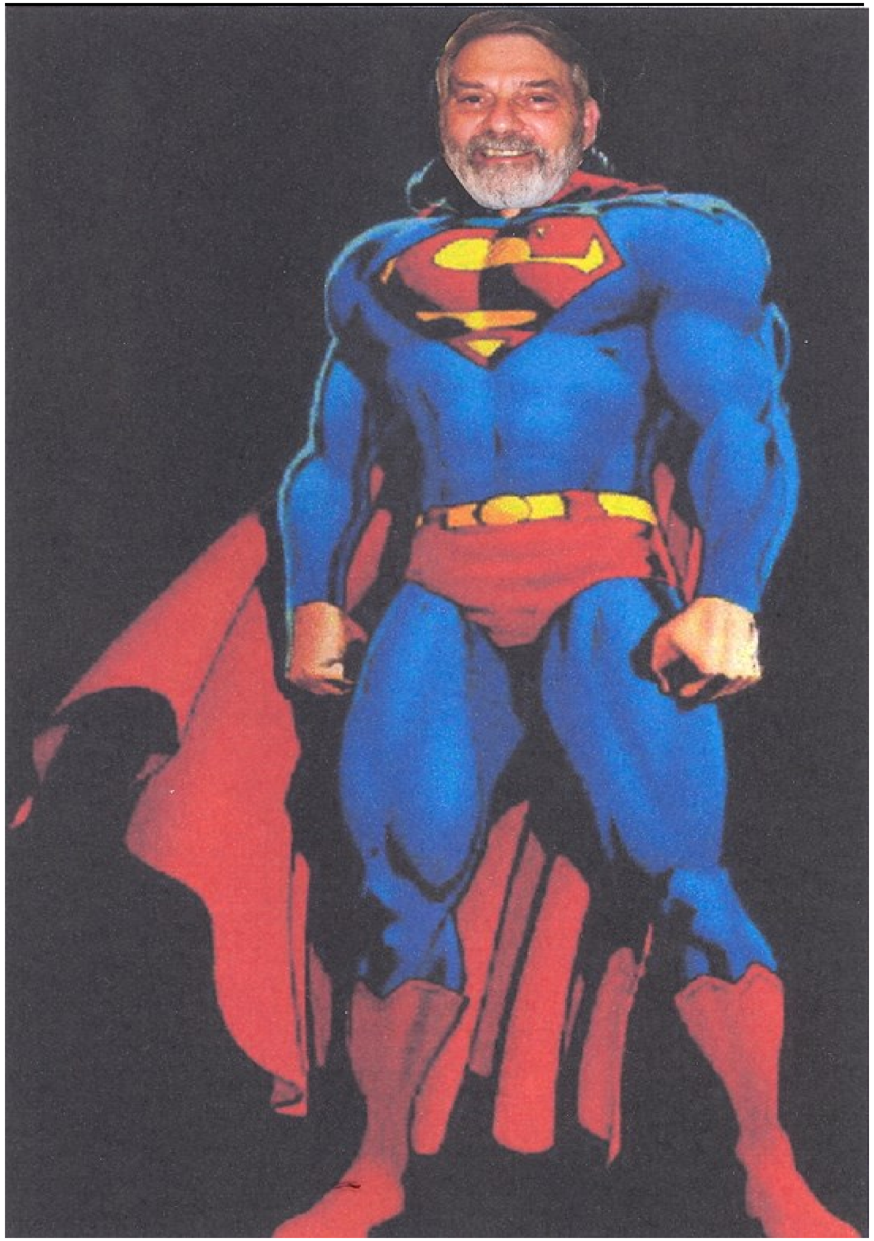
Spectrum, Zoom to Transition BW




Dilbert, is it true that DSP makes the world go around but multirate signal processing supplies the music for the ride?

Can There be any Doubt???





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