

# WHAT IS OPENHPSDR?

Open High Performance Software Defined Radio

- An international group of SDR enthusiasts.
  Developing open-source hardware and software.
  Very "High Performance" designs --- extending the state of the art of SDR in Amateur Radio.
  Having a lot of fun working with each other, challenging each other, and learning as we develop and test new technology.
- Partners (separate entities) include TAPR and Apache Labs.



OPENHPSDR TOPICS PACIFICON 2014

#### **NEW ITEMS THAT ARE NOW SHIPPING**

- WDSP New DSP library
- PureSignal Adaptive pre-distortion
- EER / ET firmware & software

**ACTIVE DEVELOPMENT** 

New Communication Protocol

**ACTIVE INVESTIGATION** 

• Direct Fourier Conversion & New Architectures



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### WDSP

Developed for our current and FUTURE openHPSDR needs

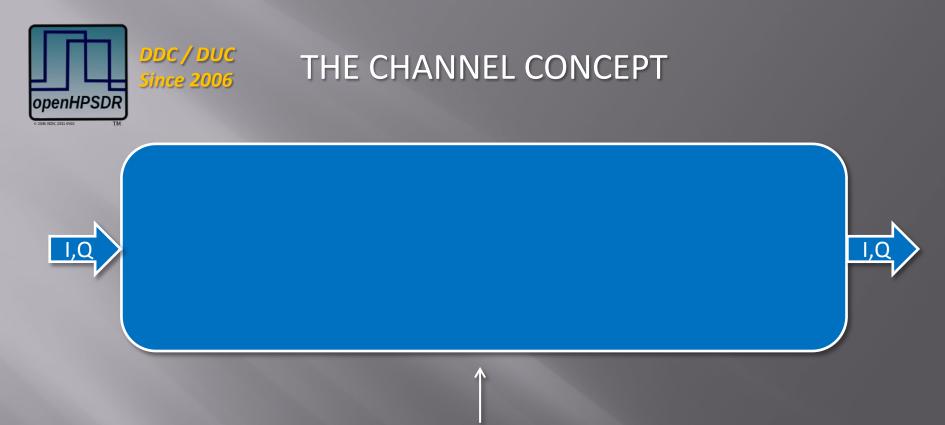
• Readily useable for other SDR projects

Many enhanced and new functions

• Open-source, GNU GPL version 2

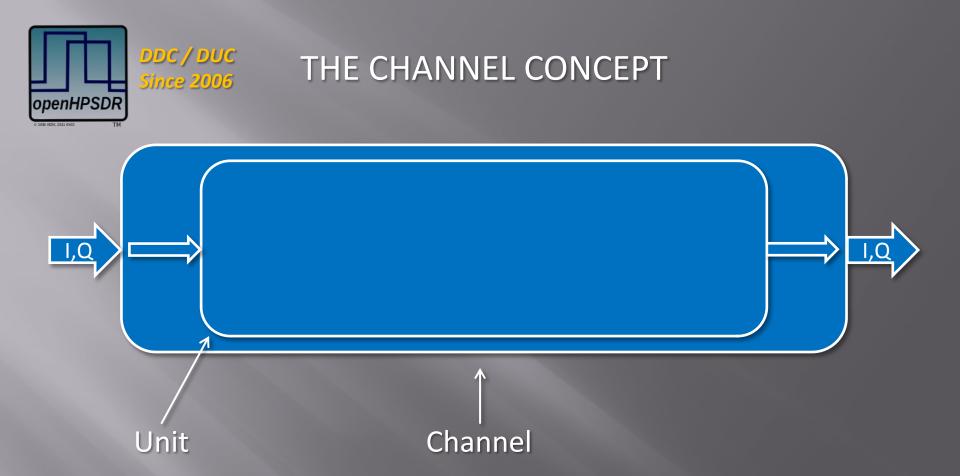
• C Programming Language – Close to the hardware & Efficient

Shipping NOW! (But, always opportunities to do more! <sup>(2)</sup>)



# Channel

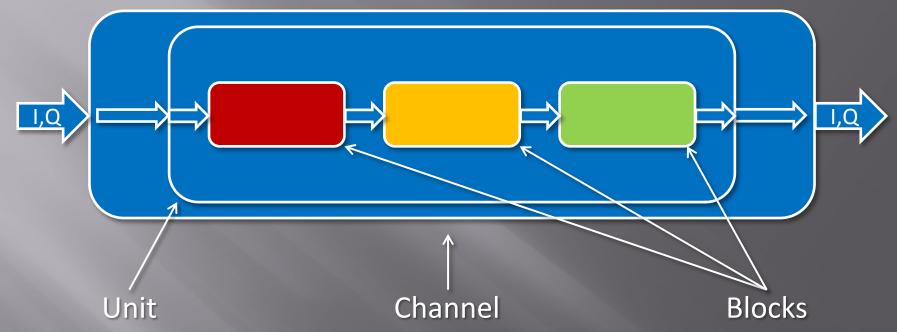
A software entity that accepts buffers of I,Q data and outputs buffers of I,Q data.



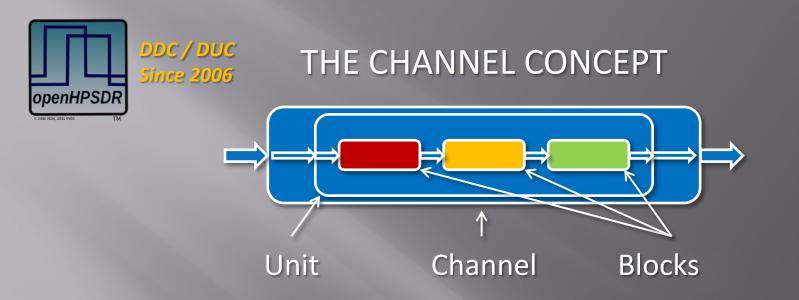
The CHANNEL provides the home for a <u>single</u> UNIT such as a receiver or transmitter.



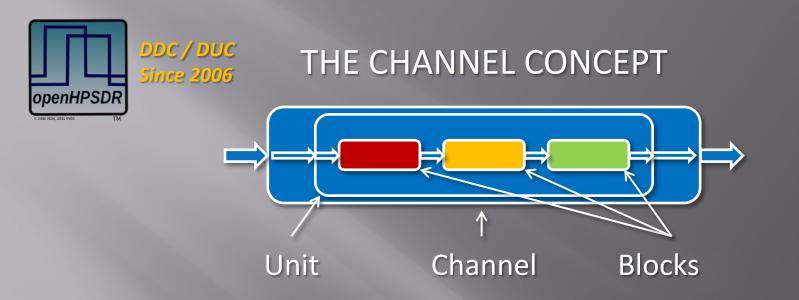
# THE CHANNEL CONCEPT



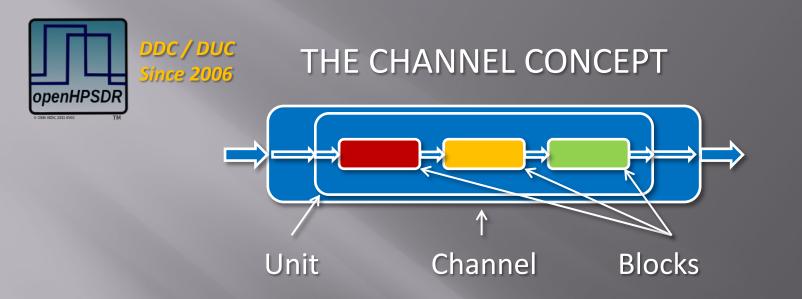
The UNIT comprises BLOCKS, each of which performs a specific function such as Filter, AM Modulator, or ALC.



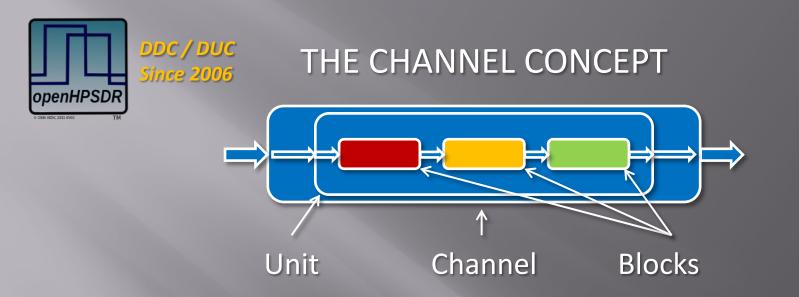
 Each CHANNEL is completely independent of all others – Channels share <u>nothing</u> – No shared data or <u>settings</u>.



 Each CHANNEL is completely independent of all others – Channels share <u>nothing</u> – No shared data or settings.
 You can have as many channels as you want.



Each CHANNEL is completely independent of all others – Channels share NOTHING – No shared data or settings.
You can have as many channels as you want.
The CHANNEL structure is <u>always exactly the same</u>, no matter what type of unit is housed within it.

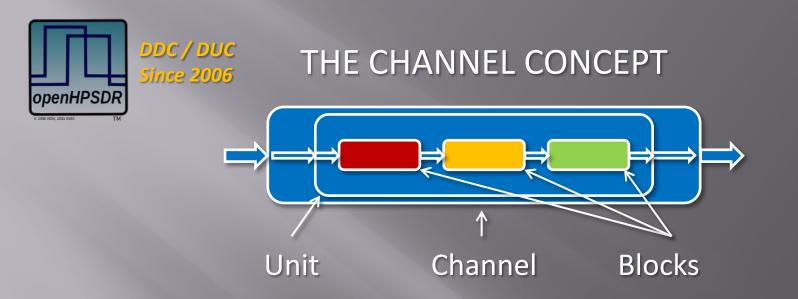


 After defining a UNIT, you can use that definition within as many channels as you want.

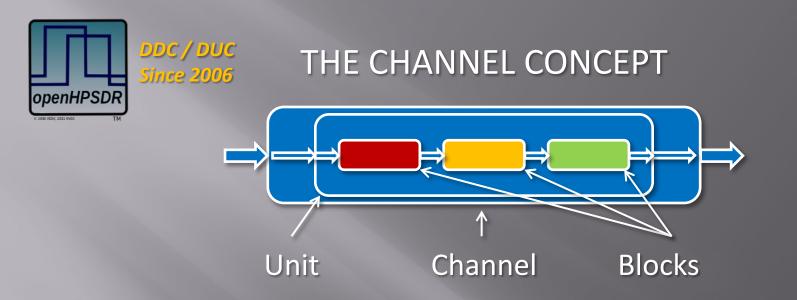
• Pre-defined units include a Receiver & a Transmitter.

It is simple to add new types of units – very uniform

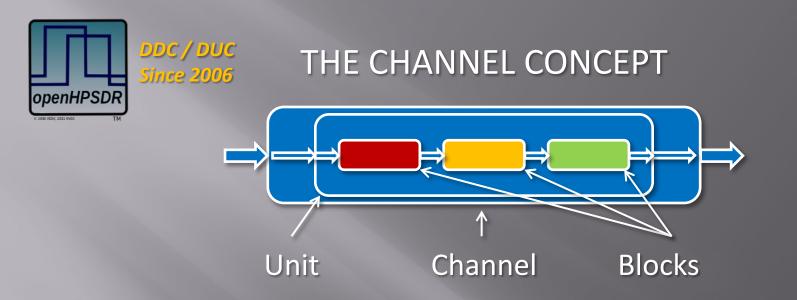
structure to "splice" them into channels.



After defining a BLOCK, you can use that definition as many times as you want within a single unit and you can use that definition in other units.
BLOCKS also have a very uniform structure.



Single CHANNEL structure, uniform UNIT structure, and uniform BLOCK structure support easy re-use of technology and more rapid development.



#### A rich assortment of BLOCKs has been included!



- Frequency-shifter (complex oscillator + mixer)
- Resampler
- Signal Generators
  - sine, pulse, two-tone, triangle, noise, sawtooth
- Adjustable bandpass filters
- AM squelch & transmit noise-gate
  - raised-cosine transitions
  - continuously variable tail length
- AM Demodulation
  - basic & synchronous modes
  - sideband selection (lower, upper, or both) true
  - phasing separation, not filtering
  - carrier stabilization



FM Demodulation

- de-emphasis
- CTCSS block
- FM Squelch
  - raised-cosine transitions
  - continuously variable tail length
- Equalizer
- continuous-gain (as opposed to gain by band)
  specify gain at any number of frequencies
  Automatic Notch Filter (LMS algorithm)

  automatic variable leak

  Automatic Noise Reduction (LMS algorithm)

  automatic variable leak



- Speech processor
  - characteristics of an RF speech processor
- AM modulator
  - zero carrier shift
  - absolute 100% modulation control available
- FM modulator
  - pre-emphasis (either before or after limiting)
  - CTCSS tones
- Preemptive NoiseBlanker
  - slew time, advance time, hang time control
- Diversity mixing
- PureSignal transmit linearity correction
- Audio Peaking Filters for CW & RTTY



#### • AGC / ALC / Audio Leveler

- ZERO overshoot (total amplitude control)
- automatic fast decay mode for transients
- hang functionality
- slope functionality (strong stations sound louder)
- get / set functions for controls on panadapter
- Patchpanel
  - select I or Q or I and Q
  - copy  $| \rightarrow Q$  or  $Q \rightarrow |$
  - mutual and separate I, Q gain controls
  - use for input select, audio pan, binaural output selection, etc.



#### • Meters

- peak, average, and gain modes
- Phase & scope display
- Panadapter / Spectrum display
  - large FFT support for weak signal
  - stitched spectra for wider display
  - adjustable frame rate (independent of sample rate and FFT size)
  - spur elimination for Cyclops spectrum analyzer
  - resamples to chosen pixel width
  - selection of window functions
  - selection of averaging modes
- AND MORE ...



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**ACTIVE DEVELOPMENT** 

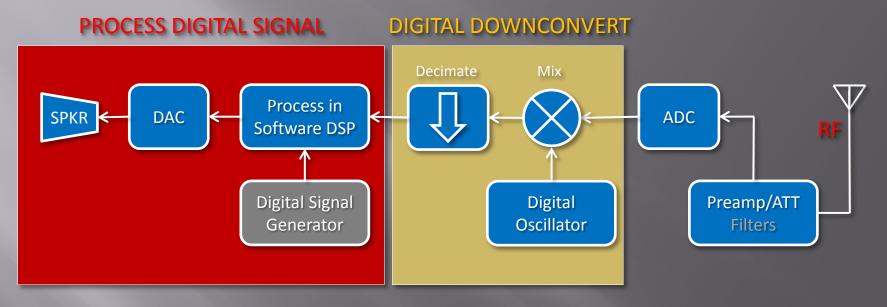
New Communication Protocol

**ACTIVE INVESTIGATION** 

• Direct Fourier Conversion & New Architectures



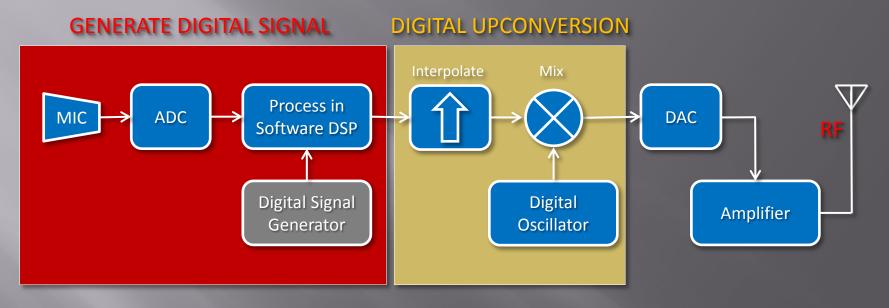
#### **DDC RECEIVER**



Mix With Complex Oscillator To Generate Baseband (0 Hz IF) Signal
 Decimate Down From The Sample Rate Of The Oscillator & ADC (122.88 Mhz)

Process The Complex Digital Signal (I,Q) To Generate Audio
 Sample rates are easily processed in software (48K – 384K)





Complex Digital Signal (I,Q) Generated From Audio Data
 Sample rates are easily processed in software (48K - 384)

• Sample rates are easily processed in software (48K – 384K)

Interpolate Up To The Sample Rate Of The DAC & Oscillator (122.88 Mhz)
 Mix With Complex Oscillator To Generate The RF-Frequency Digital Signal



DIGITAL UPCONVERSION **GENERATE DIGITAL SIGNAL** Interpolate Mix Process in DAC MIC ADC Software DSP **Digital Signal** Digital Amplifier Generator Oscillator 14.185 14.190 14.195 14.205 14.210 14.215 14.220 14.180 14.200 -20 -30 -40 -50 -60 -70 -80 -90 100 -110 -120 -130 -140



DIGITAL UPCONVERSION **GENERATE DIGITAL SIGNAL** Interpolate Mix Process in DAC MIC ADC Software DSP Digital **Digital Signal** Amplifier Oscillator Generator 14.200 14.205 14.210 14.215 14.220 14.180 14.185 14.190 14.195 -20 -30 -40 -50 -60 -70 -80 -90 100 110 -<mark>120</mark> -130 and an all and a second produce and and the second -140

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# WHY?

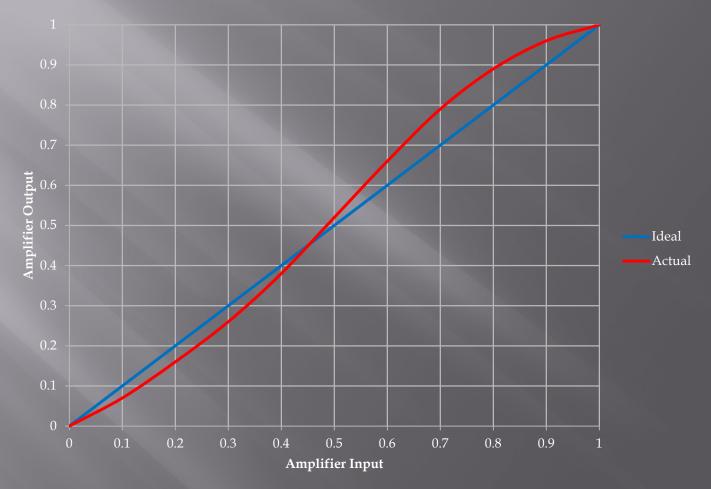


# WHY?

# Because the amplifier is NOT perfectly linear!

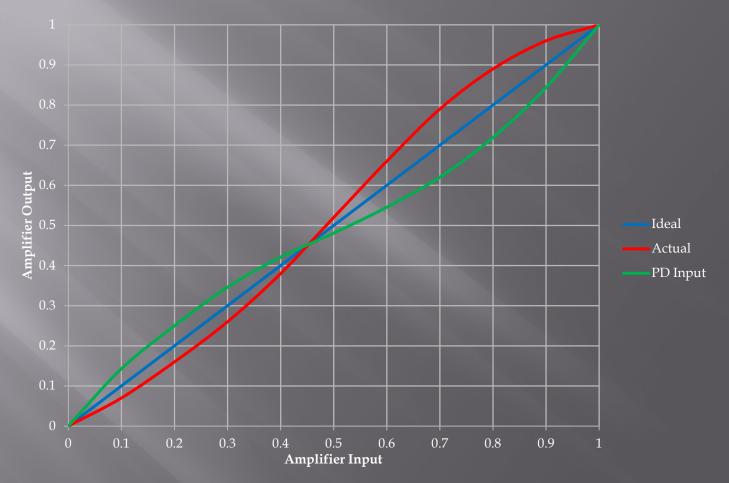


# AMPLITUDE NON-LINEARITY





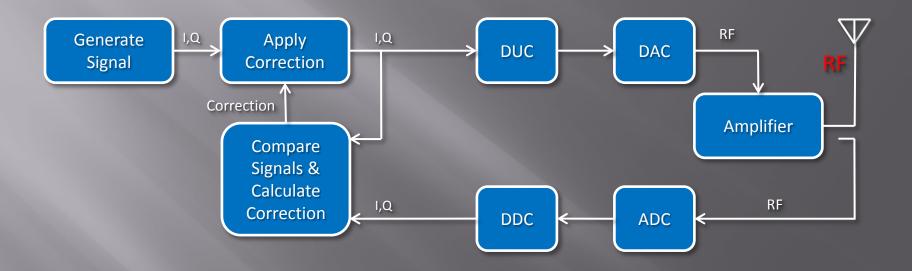
# CORRECT BY PREDISTORTION





# ADAPTIVE BASEBAND PREDISTORTION

#### Basic Concept



• Apply Correction to the out-bound signal

- Calculate Correction by Comparing the Input & Output of the Amplifier
   BASEBAND I,Q Before Up-Conversion / I,Q After Down-Conversion
  - ADAPTIVE Repeat the process to Adapt to Changing Conditions



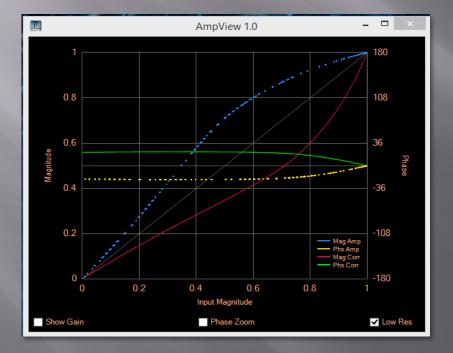
# PURESIGNAL RESULTS Clyde, K2UE







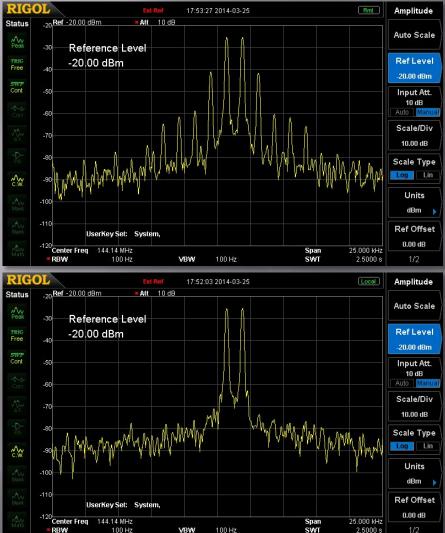
ANAN Low-Pwr Xvtr Output
Full-duplex Transverter
1200W 2M Amplifier



2M Amplifier is VERY non-linear
LDMOS, Very low memory effects
Should be very correctable!

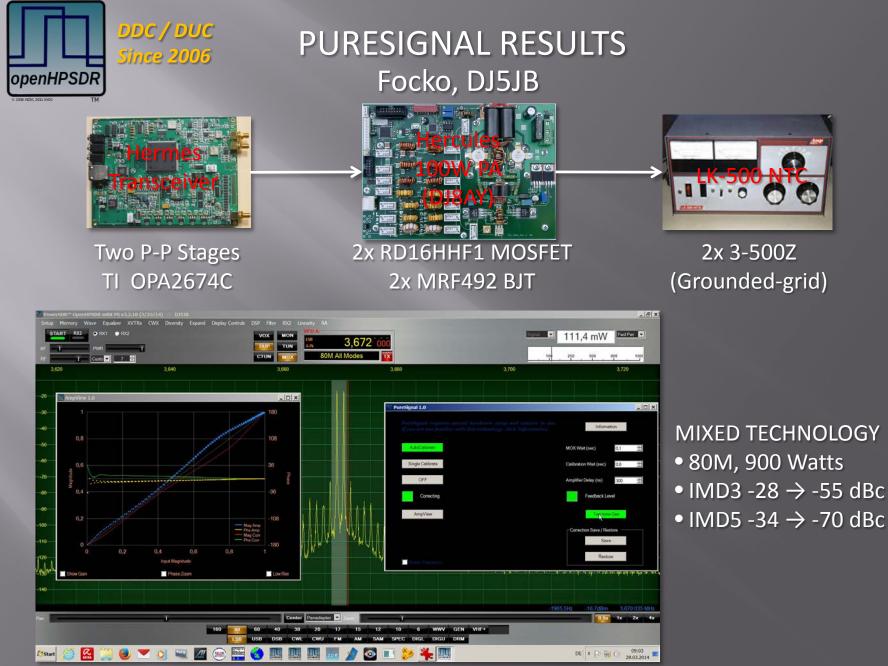


### PURESIGNAL RESULTS Clyde, K2UE



# PureSignal OFF IMD3 ~ -16dBt

PureSignal ON
IMD3 ~ -48dBt





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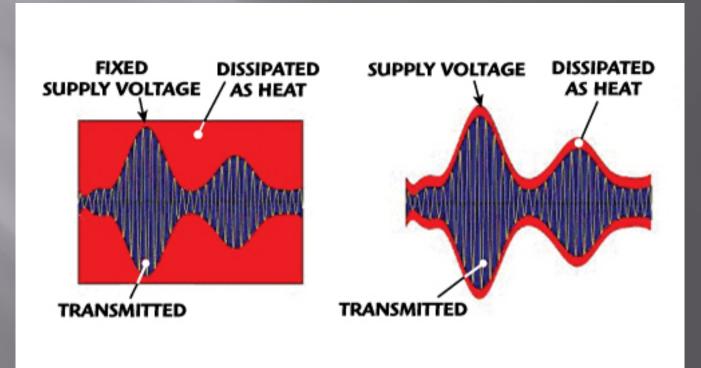
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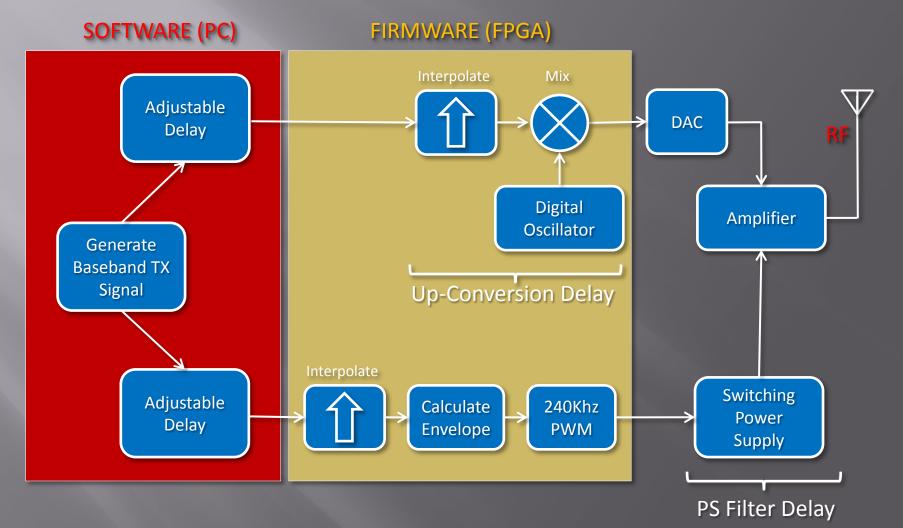


#### HIGHLY-EFFICIENT POWER AMPLIFIERS EER (Envelope Elimination & Restoration) ET (Envelope Tracking)





#### EER / ET IMPLEMENTATION





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#### NEW PROTOCOL FEATURES

• Each ADC can feed multiple DDCs.

- Each DDC can output a different sample rate.
- Completely independent DDCs.
- Multiple Synchronous DDCs
  - DDCs on separate ADCs can synchronously combine
- Multiple Synchronous DUCs
  - Beam forming
- DDC feeding DUC linear translator

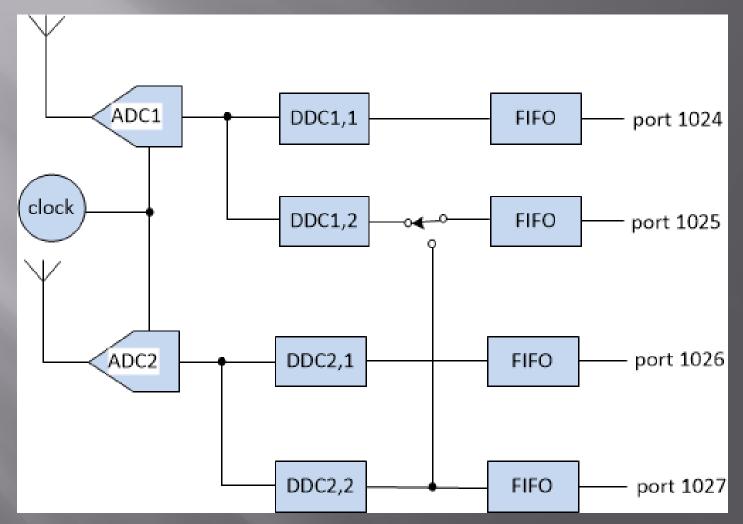


#### NEW PROTOCOL FEATURES

- Based on *streams*
- Stream format defined when opened
- Standard stream types initially defined
- New stream types easy to define and add
- Use UDP port numbers to identify streams
- Port number assigned when stream opened



#### NEW PROTOCOL FEATURES



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Credit – Phil Harman, VK6PH



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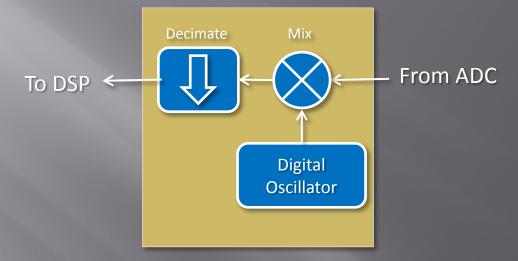
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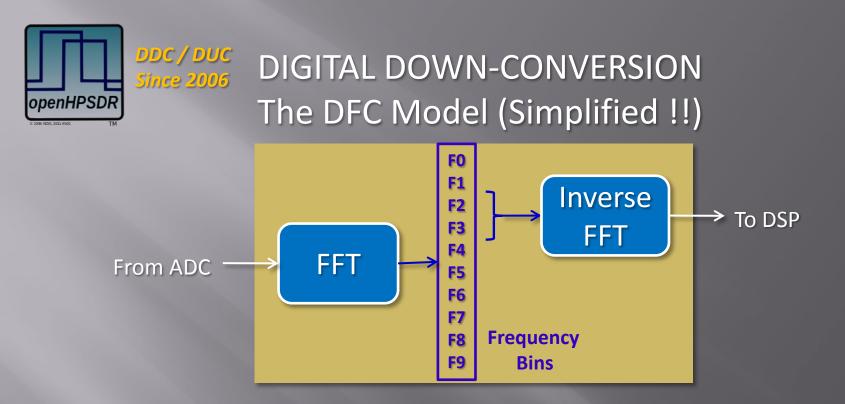
Direct Fourier Conversion & New Architectures



### DIGITAL DOWN-CONVERSION The Current Model



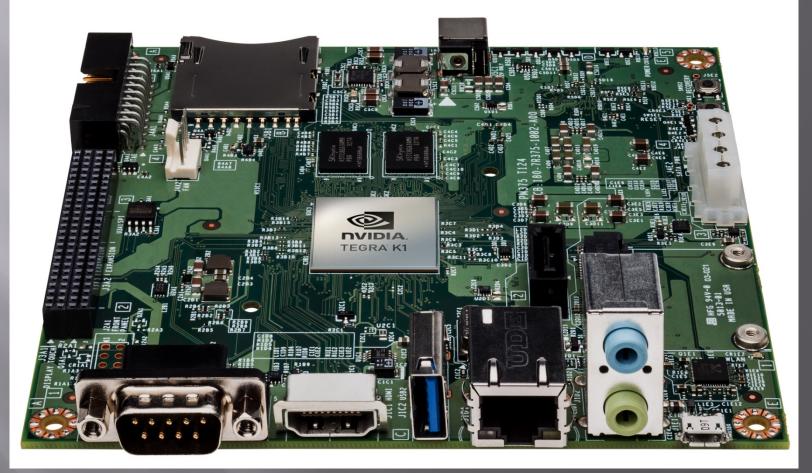
- Done in the time domain, in an FPGA
- Each DDC requires replicating the FPGA resources
- Each different frequency slice requires another DDC
- FPGA programming is less productive than software



- Done in the frequency domain, in an SBC or PC
- All DDCs use the same forward transform (the big one)
- Each different frequency slice requires an IFFT
- Programming is in software, e.g., in C



#### DDC / DUC Since 2005 DIGITAL DOWN-CONVERSION The DFC Model



Nvidia Jetson TK1: Quad-core ARM, 192 Cuda Cores, \$ 192



#### NEW ARCHITECTURE Future Possibilities

Single

Board

Computer

Simplified SDR Hardware



# QUESTIONS?