

# The Digital Analog of Analog for network transmission of multimedia - BRIMS - HP 1997

---

Jon Crowcroft,

<http://www.cl.cam.ac.uk/~jac22>

## The Digital Analog of Analog -

### Joint Receiver, Channel and Source Coding for Re-Routeable Multimedia Flows

Integrated Services in the Internet have tried to mimic B-ISDN .

This is a mistake of the greatest order - the strength of IP lies in soft state, dynamic routing, self-organisation and so on.

The design of flow types for IntServ should recognize this. To retain the advantages of IP, we should abandon the hard bounds devised by Parekh and re-design multimedia applications (and partially redesign the routing and forwarding model of an IP router (or switch)), to provide statistical properties.

In this talk, I look at one (extreme) way that we might start to consider such a re-design. This idea was triggered by reading some of David Tennenhouse's work on SpectrumWare (<http://www.tns.lcs.mit.edu/SpectrumWare/home.html>)

The basic idea (as per the title) is to re-visit the analog world for ideas when designing both the sender and receiver coders, and the channel -

## The Classic Sender-Channel-Receiver Picture:



Video Grabber+  
Coder+  
Packetizer

The Internet!

Playout Estimator/Buffer+  
Decoder+  
D2A device/Framebuffer...

or

Camera/Mike

Cable

TV or Amp+Speaker

Now consider noise on the channel.....and what it does to each type of signal!

## The Classic Sender-Channel-Receiver Picture:



How is signal on channel modified?

1. In analog network

- i) by attenuation (transfer function depends on frequency)
- ii) by interference ( random changes introduces to signal by adding in some signal)

2. In Internet

- i) by packet loss caused by queue overload (congestion)
- ii) by packet delay caused by bursts of our, and other sources

How can we modify coding scheme to look like analog signal, and queueing scheme to look like analog channel?

How can we modify routing to look like graceful degradation?

## The Classic Sender-Channel-Receiver Picture:



1. Remove Artifice of periodicity (packet boundary) in source - to permit re-packetization in intermediate nodes -

At one level, this is basically FEC or Scalable Coding with packet marking (ghanbari, garrett etc), but we could go further and make queue more like CDMA!

2. Remove artifice of periodicity in grabber (the frame grab model of source is wrong - we want object/delta + conditional replenishment model, to reduce the variation....)

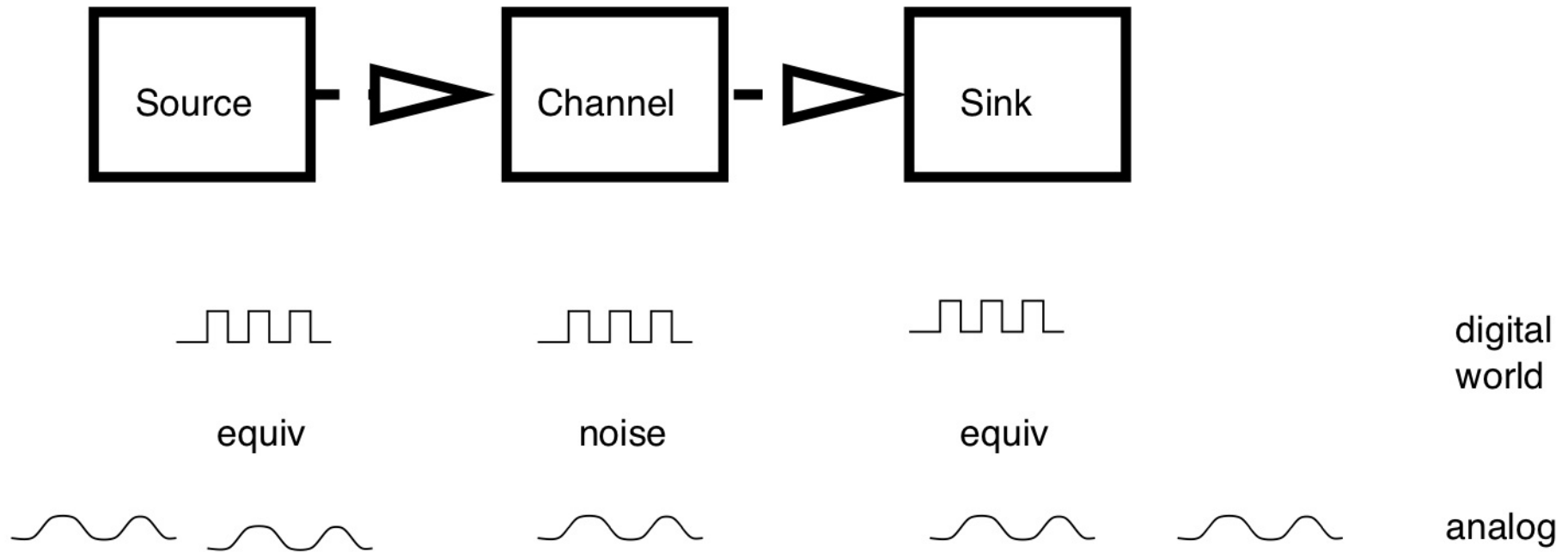
3. Remove size of step function on re-route for flows by

i) re-route all flows that are not changing rapidly (easy for marked packets in flow....if the marking is not binary, but is progressive - i.e. the more change in source, the higher the don't re-route value is....)

ii) constrain re-routes to not alter delay too much - also easy if

a) queues are short and b) only care about e2e (propagation) delay difference

## The Modified Sender-Channel-Receiver Picture:



So model loss and re-route as noise -

loss is caused by "attenuation" so how do we design a signal with max quality info in the low power.....?

To summarise:

Internet works due to low cost approach to traffic requirements

How to design continuous media streams so they let us keep the maximum benefits of this?

How to do ensemble re-routing ? easy.....probably..... some ideas from thermodynamics might help (wave/particle v. flow/packet duality) - model network as lattice gas (node occupancy = queue size) and flow as diffusion maybe?

Need to think about what new style arrival process +WRED or whatever looks as a noise function!