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QoS Tips and Tricks for VoIP Services:

Delivering reliable VoIP Services

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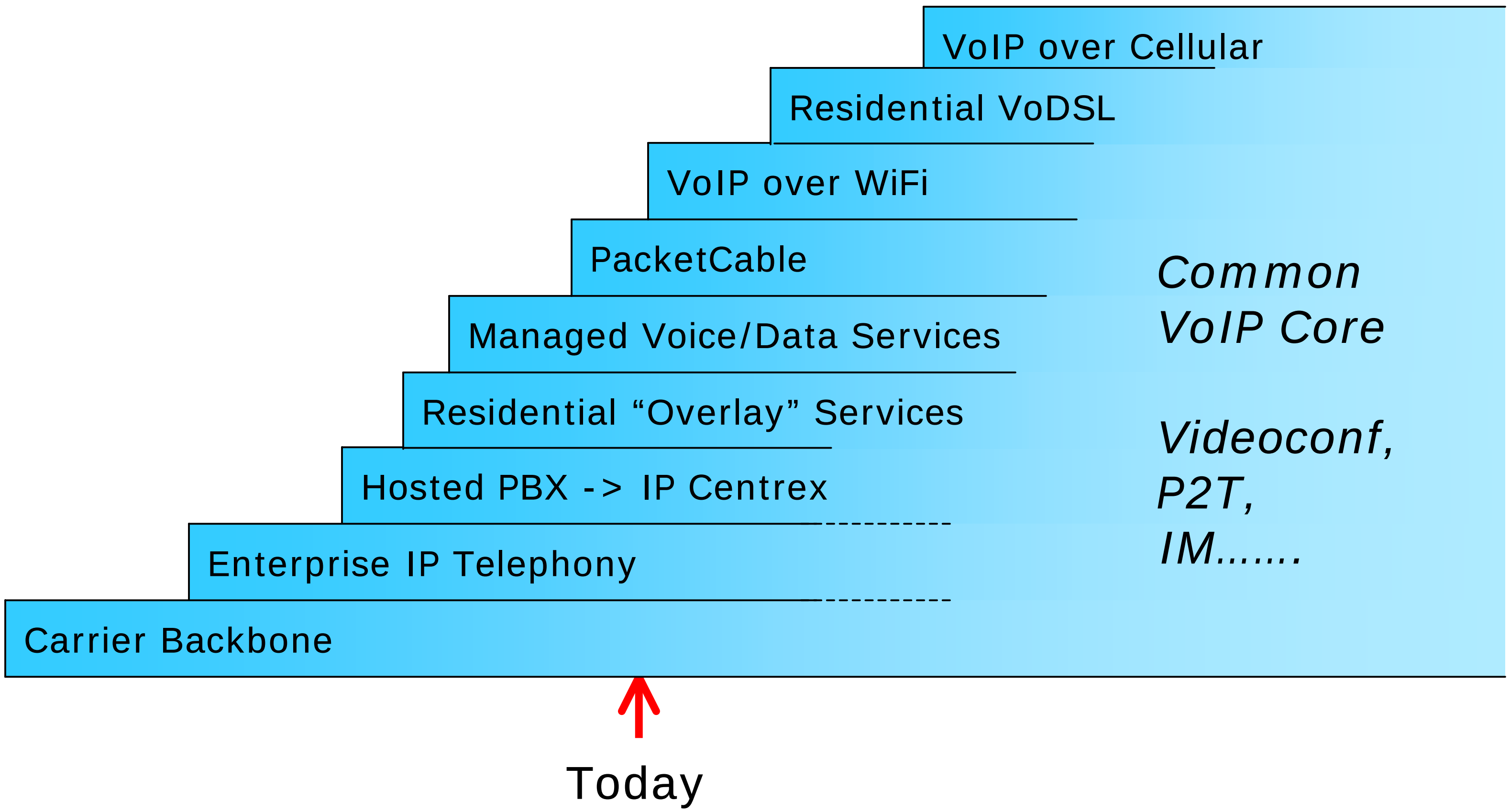
Objectives

- Clear understanding of:
  - typical problems affecting VoIP service quality
  - industry best practices for monitoring quality and diagnosing problems
  - approaches to mitigating quality problems
- Guidance on:
  - Performance management architecture
  - Pre-deployment and requirements definition
  - Deployment/ integration testing
  - Operations

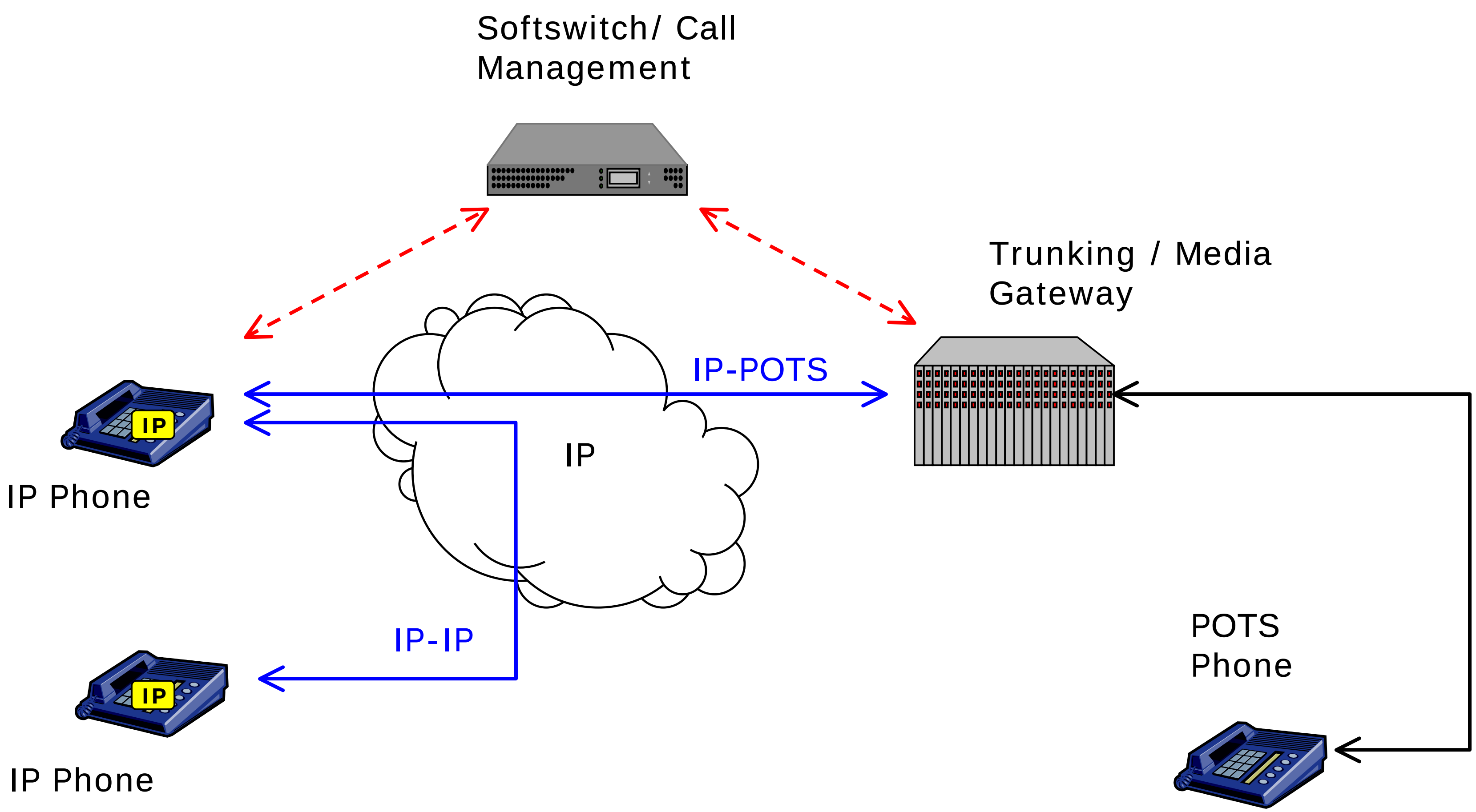
Outline

- Brief VoIP outline/ update
- Problems affecting VoIP/ VoWiFi performance
- Tools for Measuring and Diagnosing Problems
- Performance Management Architecture
- Approaches to improve performance
- Planning guidelines

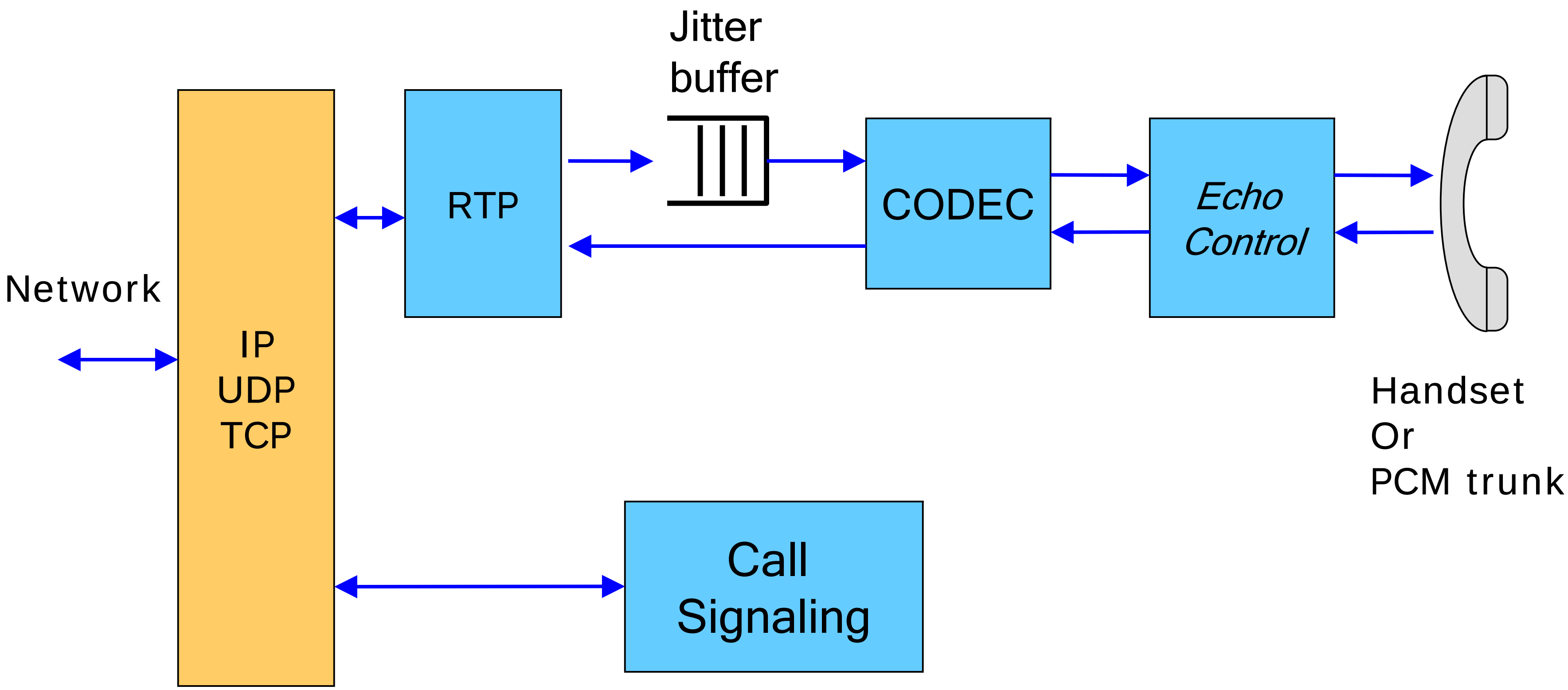
Voice over IP - One technology - many services



Components of an IP Telephony System



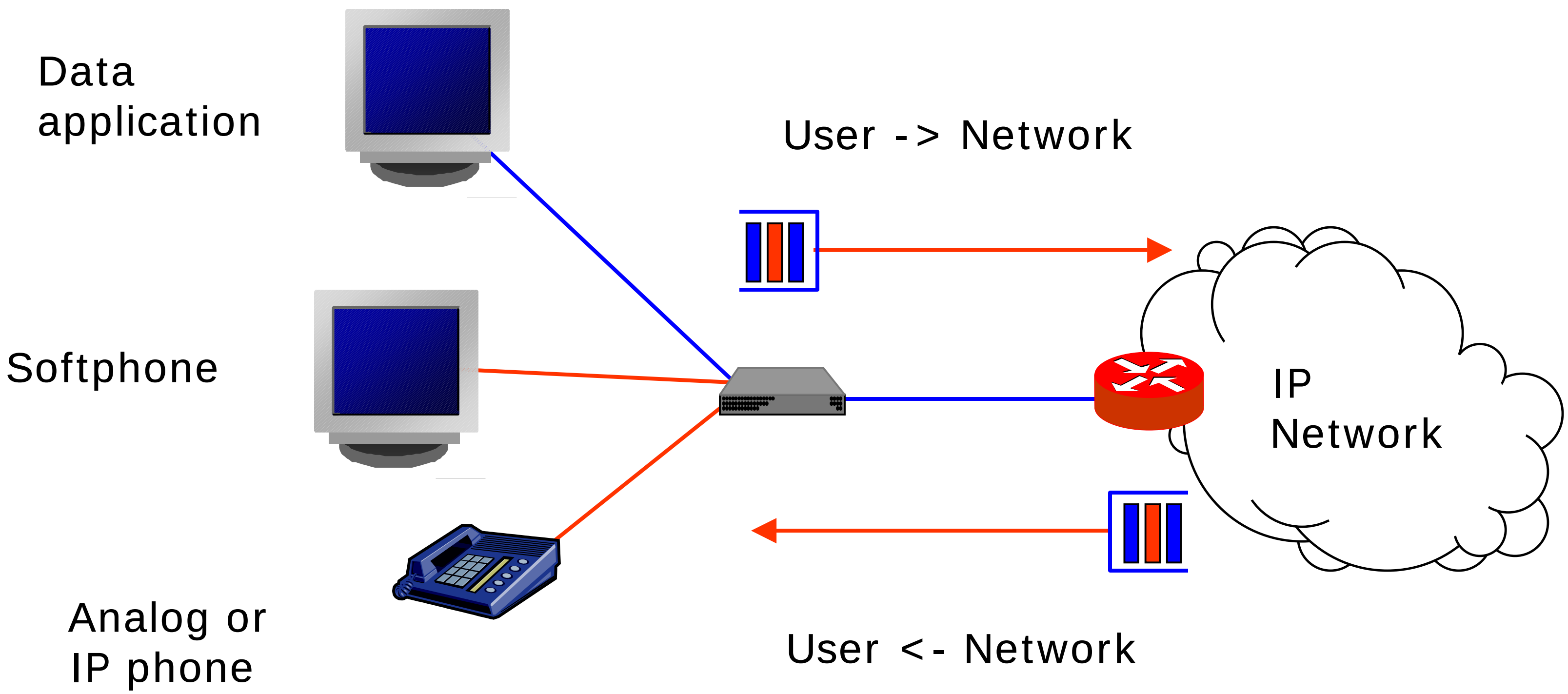
Inside an IP Phone or Gateway



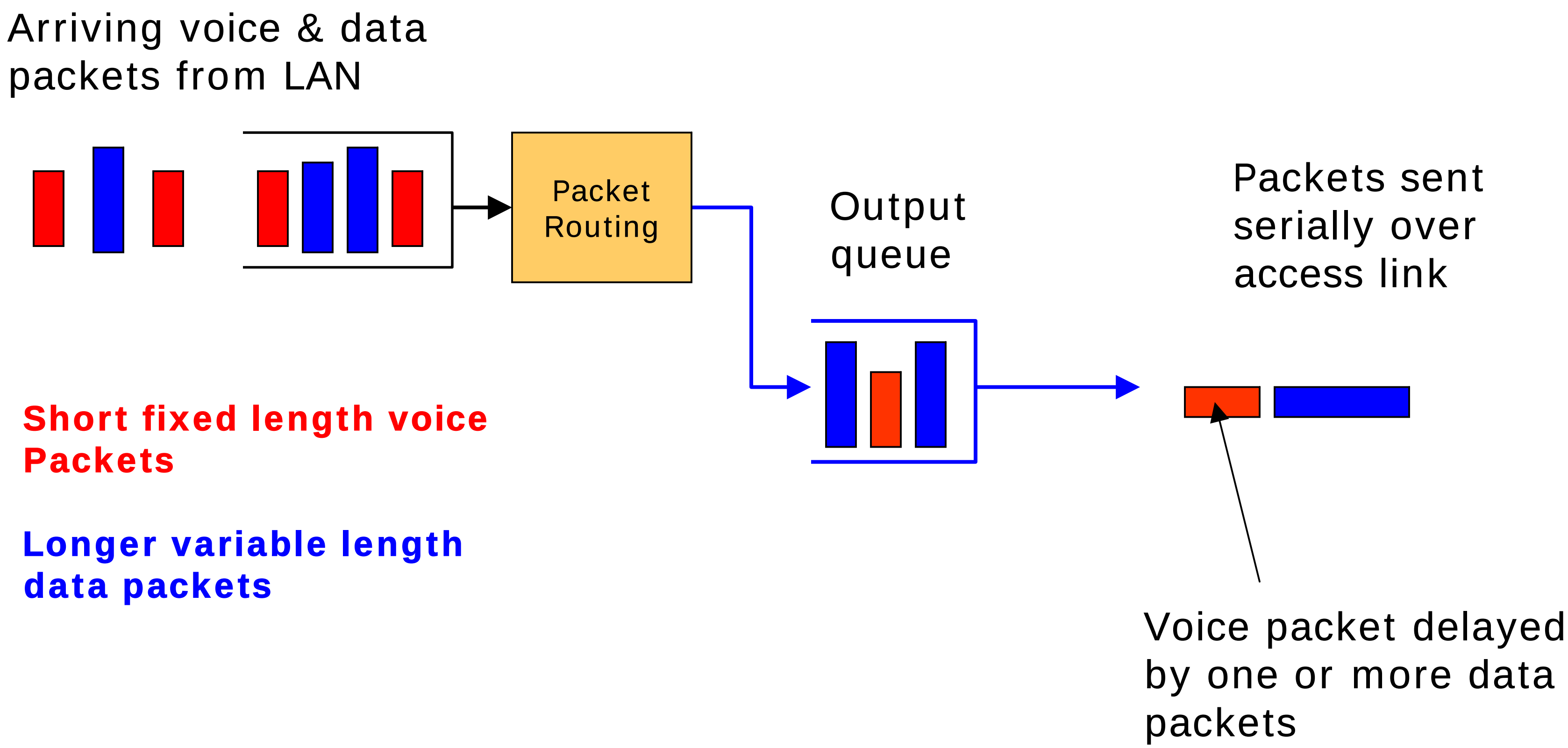
Call Quality Problems

- Packet Loss
- Jitter (Packet Delay Variation)
- Codec distortion
- Delay (Latency)
- Echo
- Signal Level
- Noise Level
- .. and various combinations thereof!!

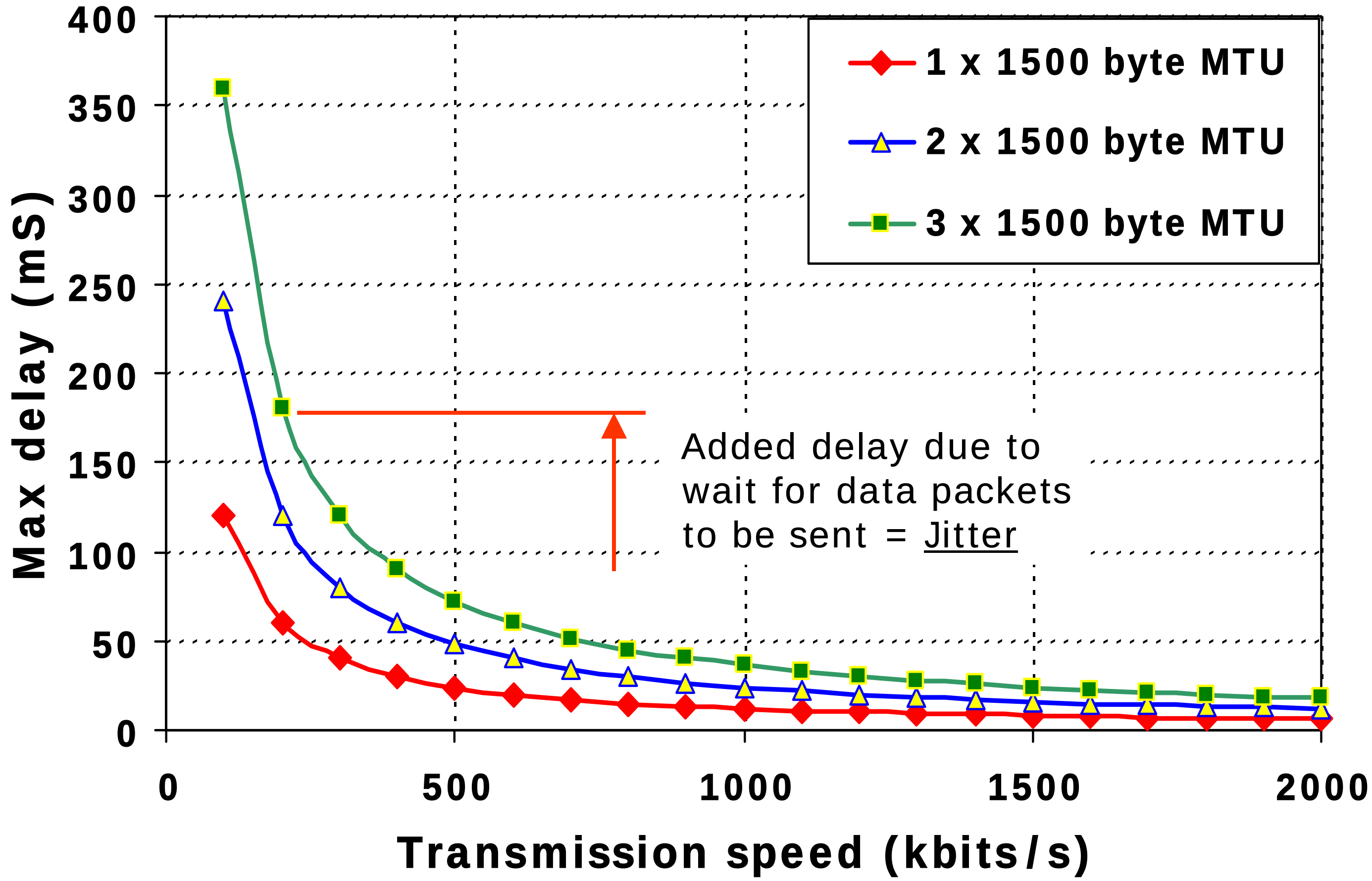
Residential VoIP scenario



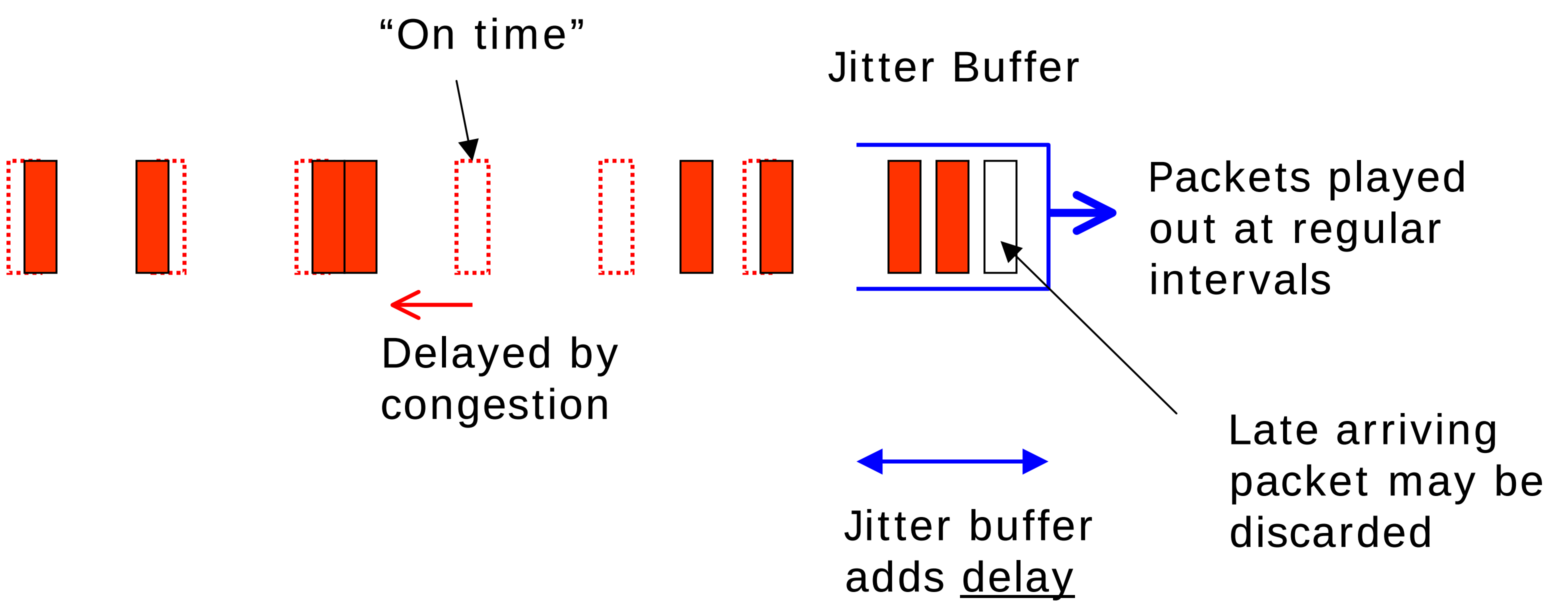
Anatomy of a Router or Residential Gateway



Effects of queuing delays in access routers



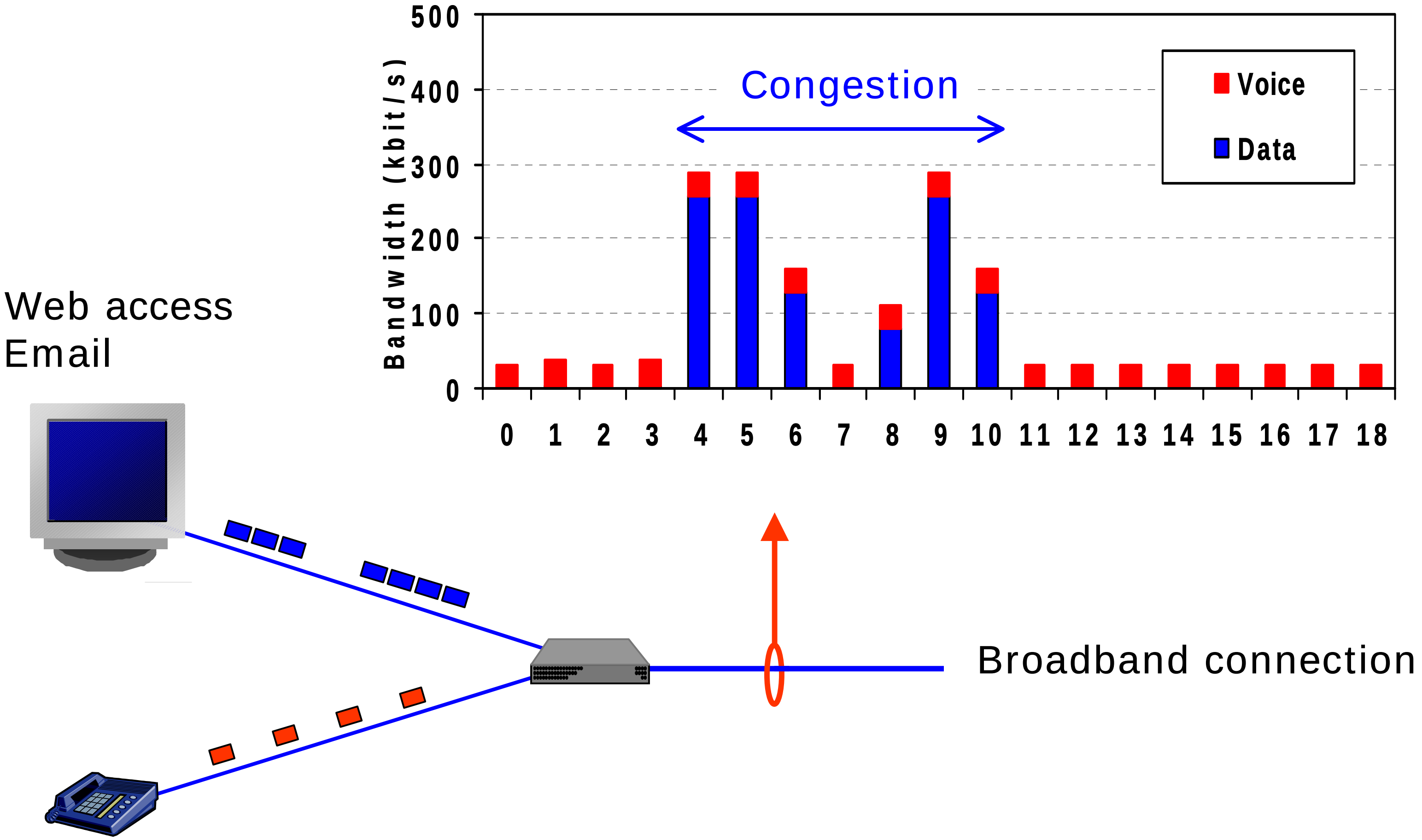
Jitter leads to Packet discard



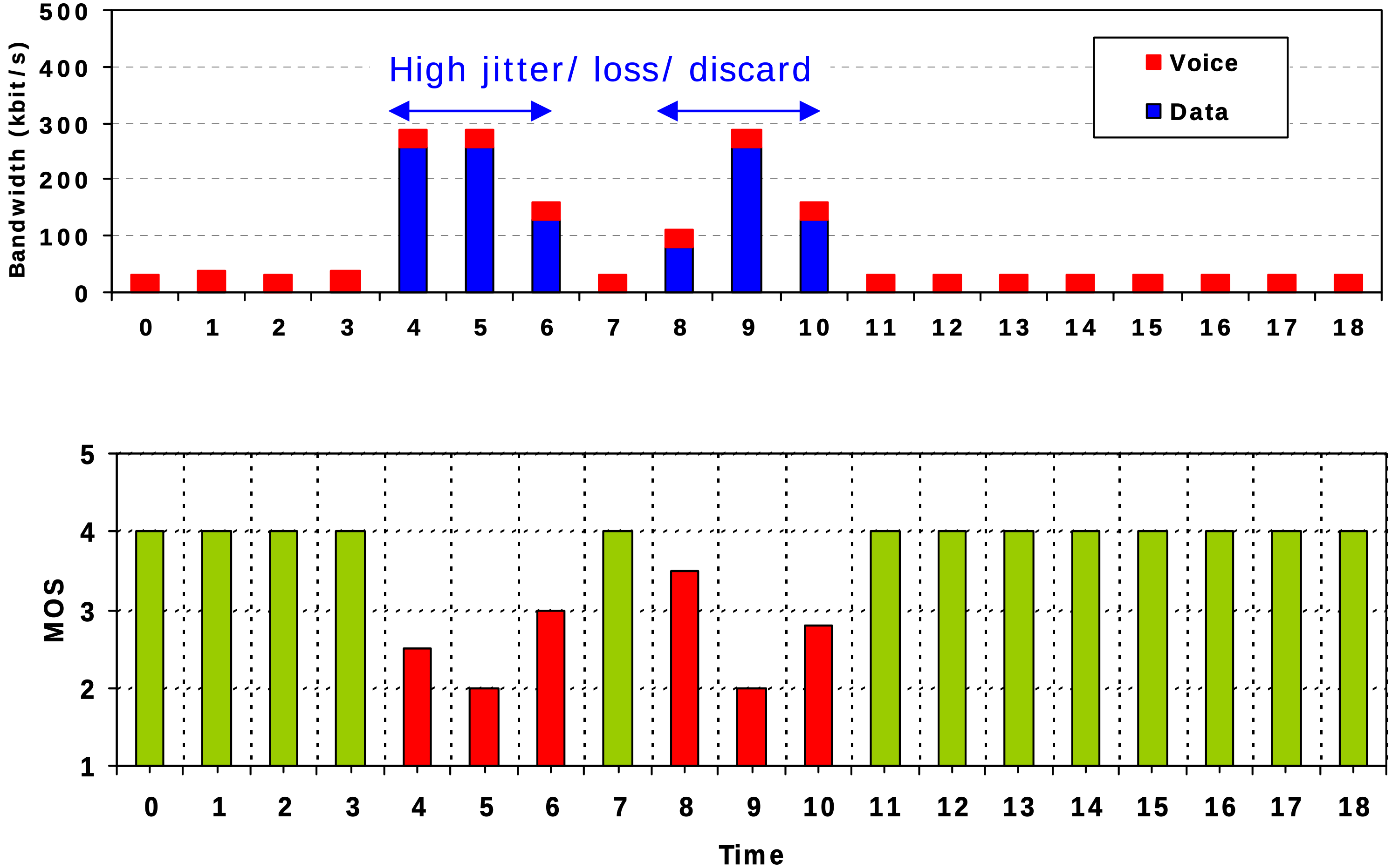
Packet Loss vs Jitter

- Low levels of jitter absorbed by jitter buffer
- Higher levels of jitter cause adaptive jitter buffer to grow - increases delay
- Very high levels of jitter lead to packets being discarded
- If packets are discarded by the jitter buffer as they arrive slightly too late they are regarded as “discarded”
- If packets are discarded within the network or arrive extremely late they are regarded as “lost”

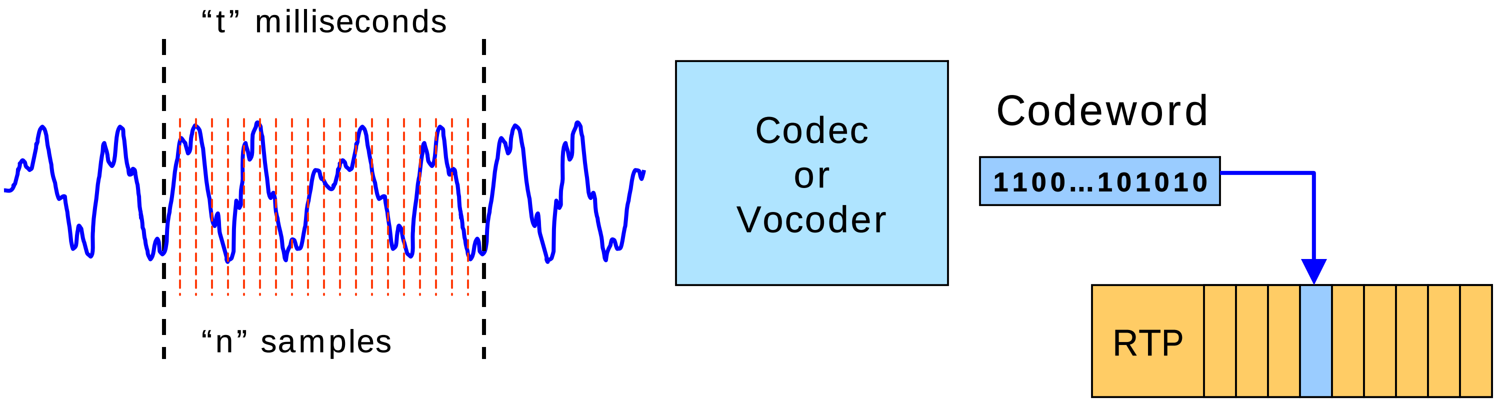
Data traffic is BURSTY



Leads To Time Varying Call Quality



Codecs and Packet Loss Concealment Algorithms



One or more voice frames
Or codewords per packet

Translates a block of speech samples to a codeword/ frame

G.711 -> 1 sample -> 8 bit codeword

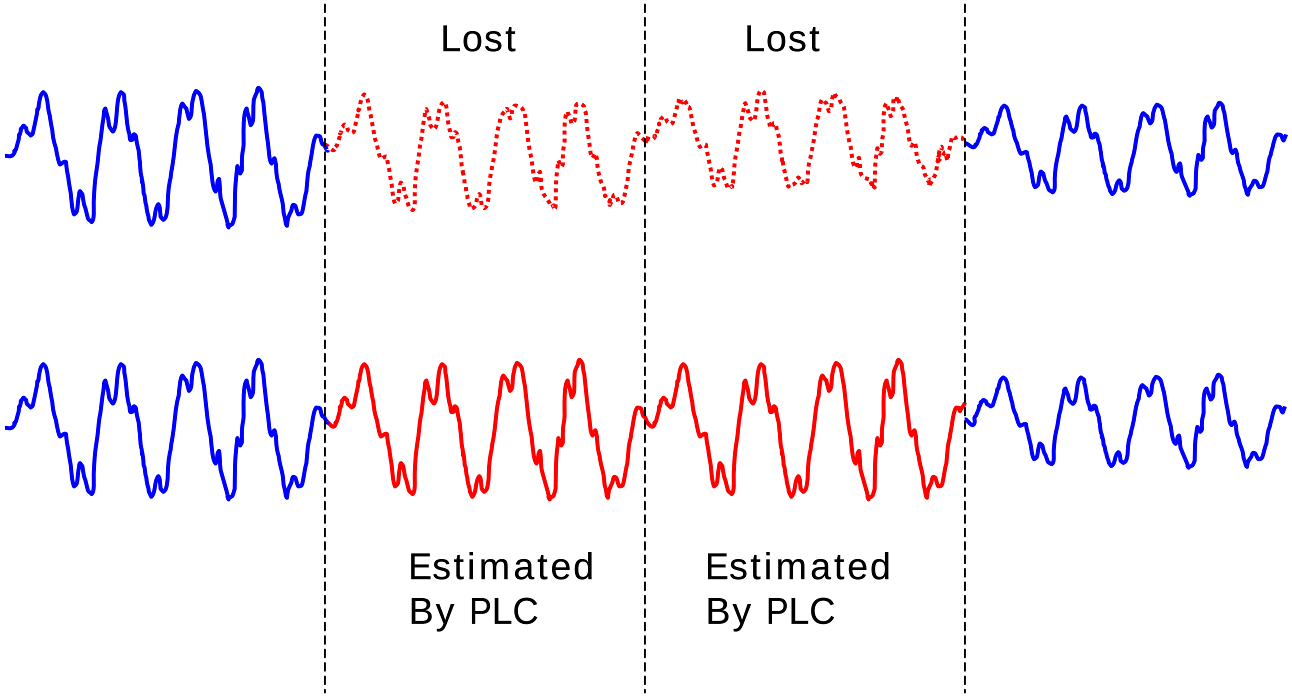
G.729A -> 80 samples -> 80 bit codeword/frame

G.723.1 5.3kbps -> 240 samples -> 160 bit codeword/frame

Codec performance

	Frame	Bitrate	Effective bitrate	MOS
G.711	10mS	64k	98k	4.1
G.711	20mS	64k	81k	3.6
G.723.1	30mS	5.3k	16k	3.6
G.729A	10mS	8k	42k	3.9
G.729A	20mS	8k	25k	3.9

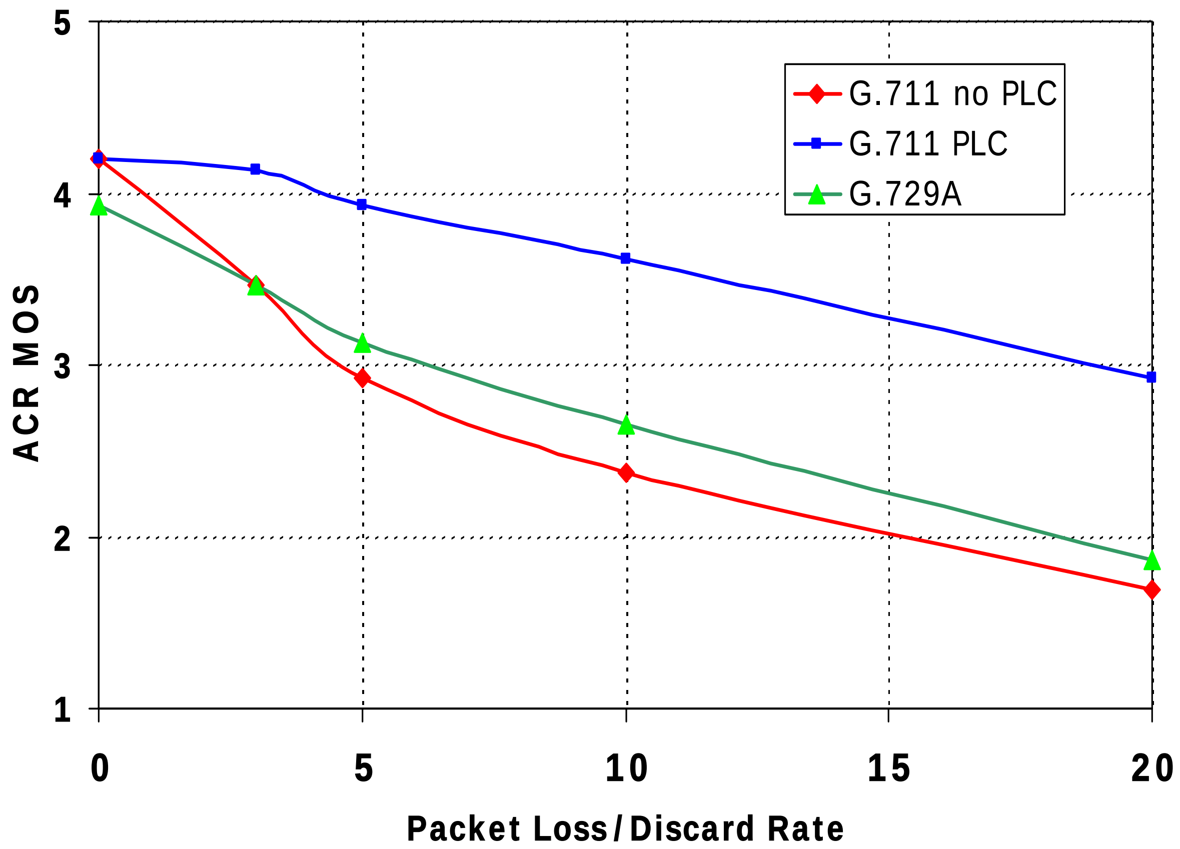
Packet Loss Concealment Algorithms



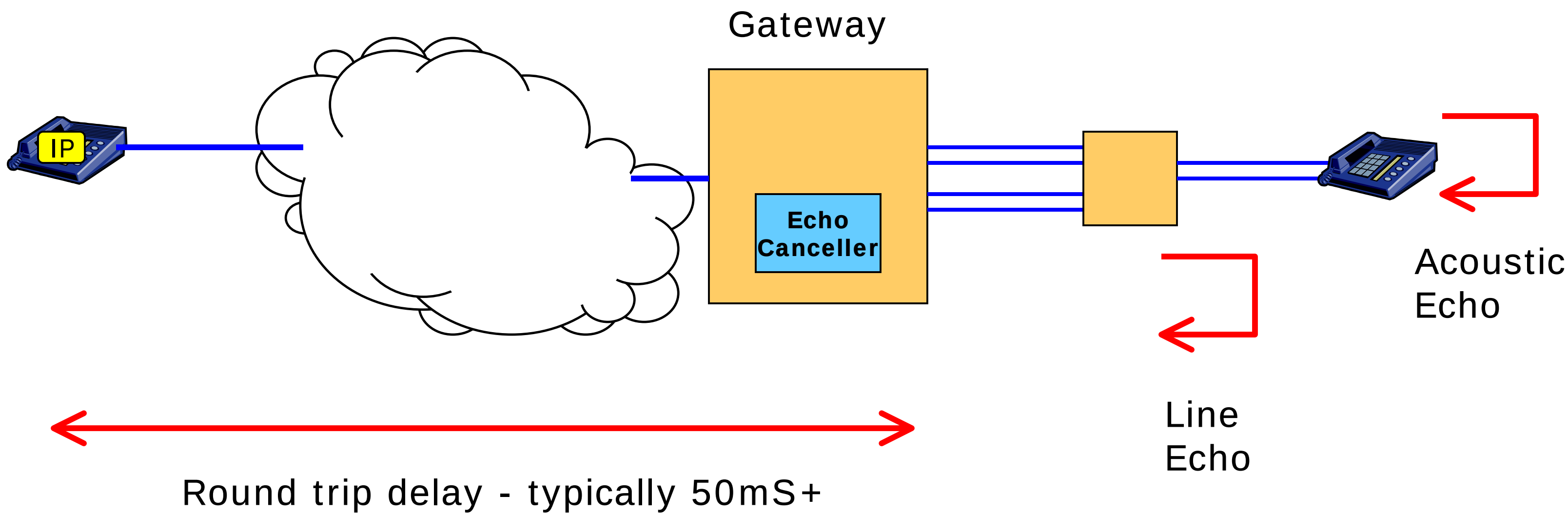
Problems

- PLC estimate is based on last frame, only good for approx 20-30mS
- Discontinuity when next frame received

Codec performance



Echo Problems



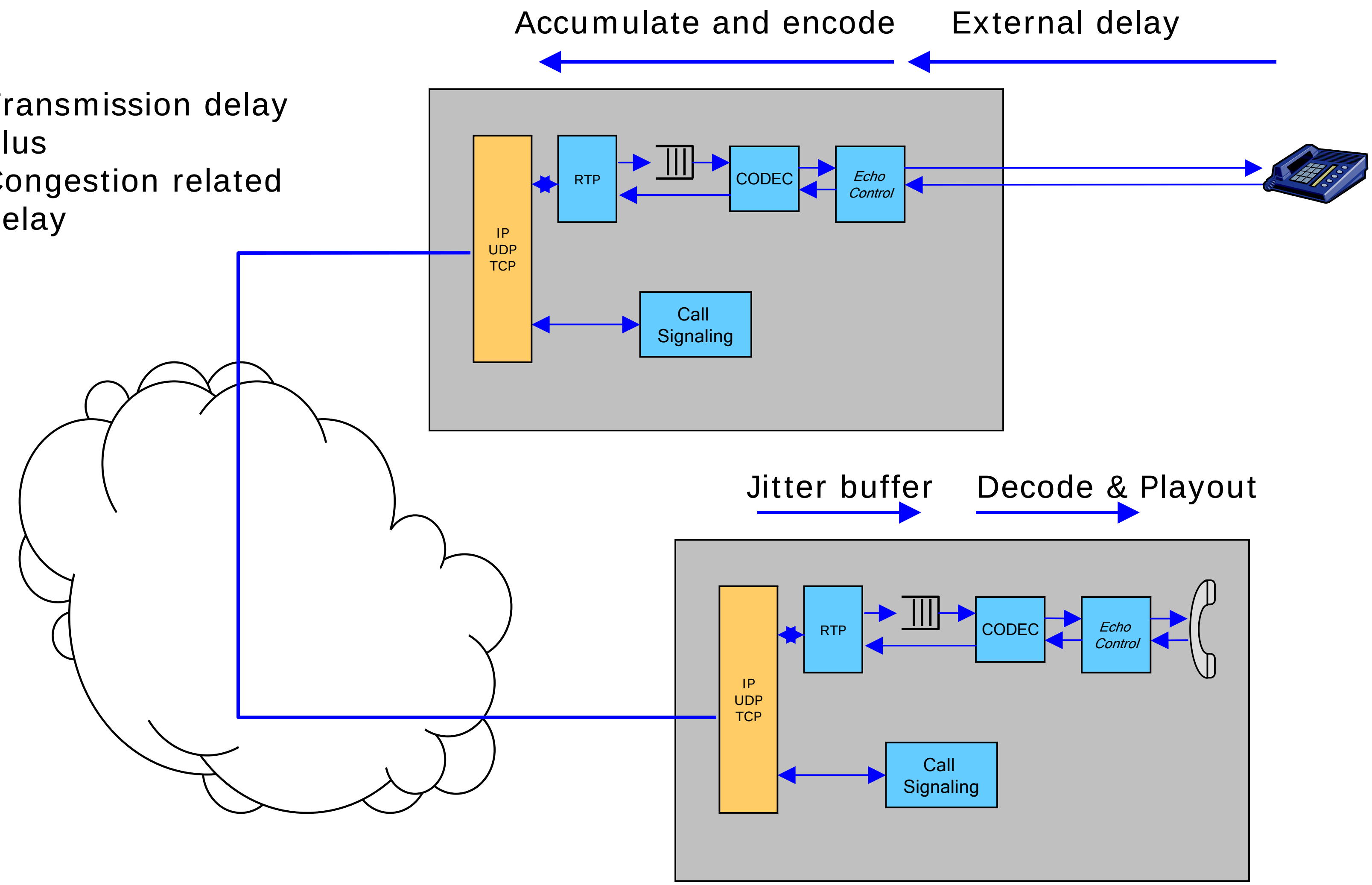
Additional delay introduced by VoIP makes existing echo problems more obvious

Echo problems

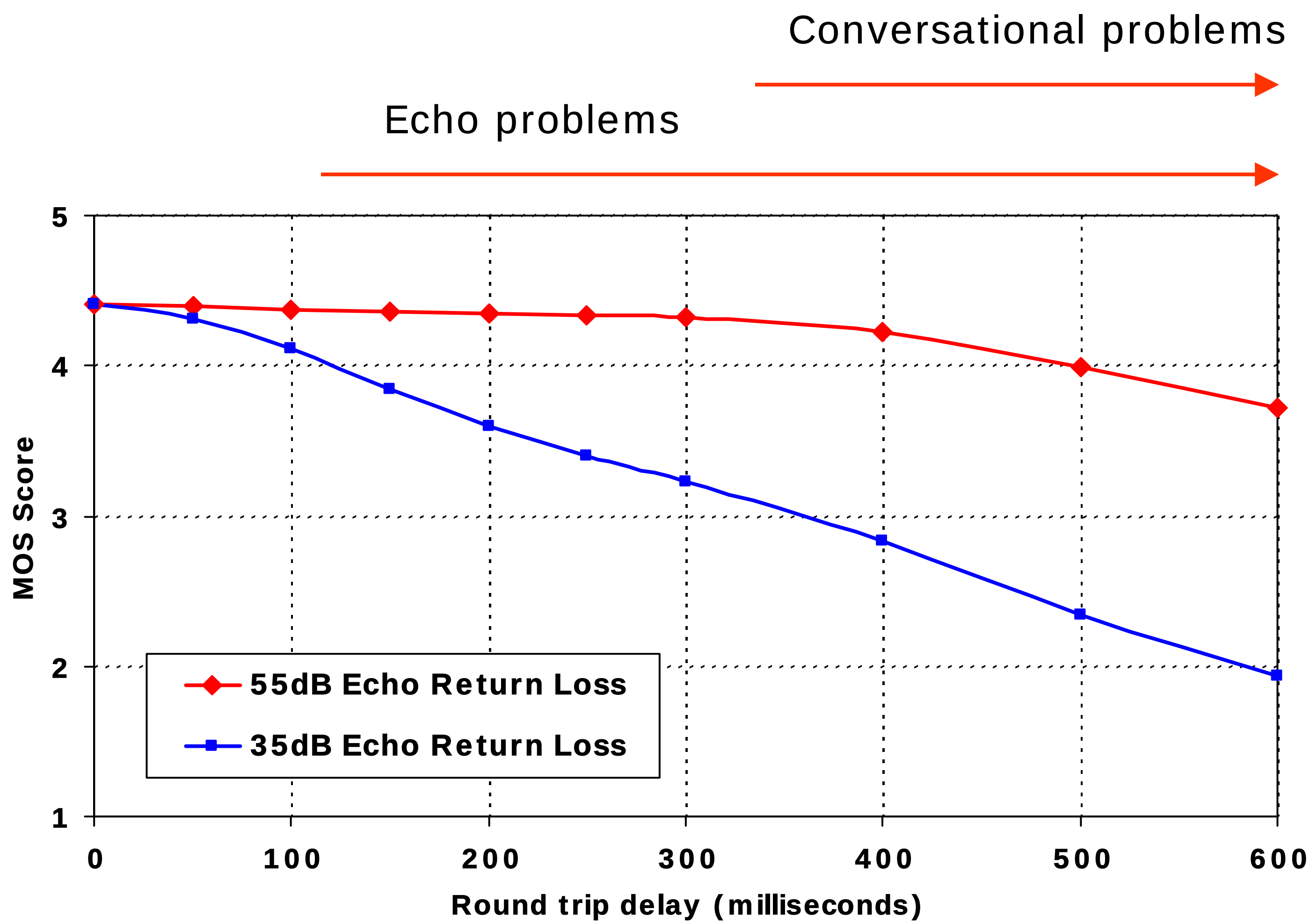
- Echo with very low delay sounds like "sidetone"
- Echo with some delay makes the line sound hollow
- Echo with over 50mS delay sounds like…. Echo
- Echo Return Loss
  - 55dB or above is good
  - 25dB or below is bad

Causes of Delay

Transmission delay
plus
Congestion related
delay

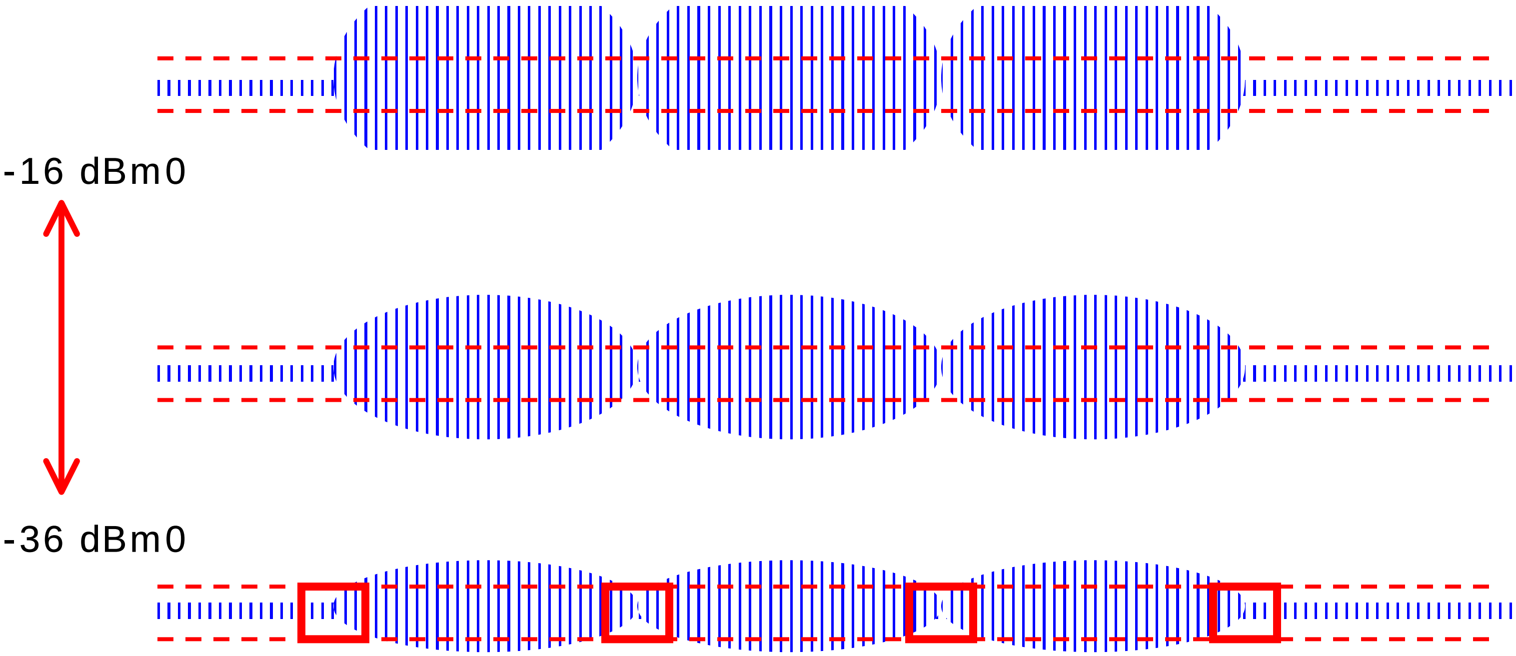


Impact of Delay



Signal Level Problems

Amplitude Clipping occurs -- speech sounds loud and “buzzy”

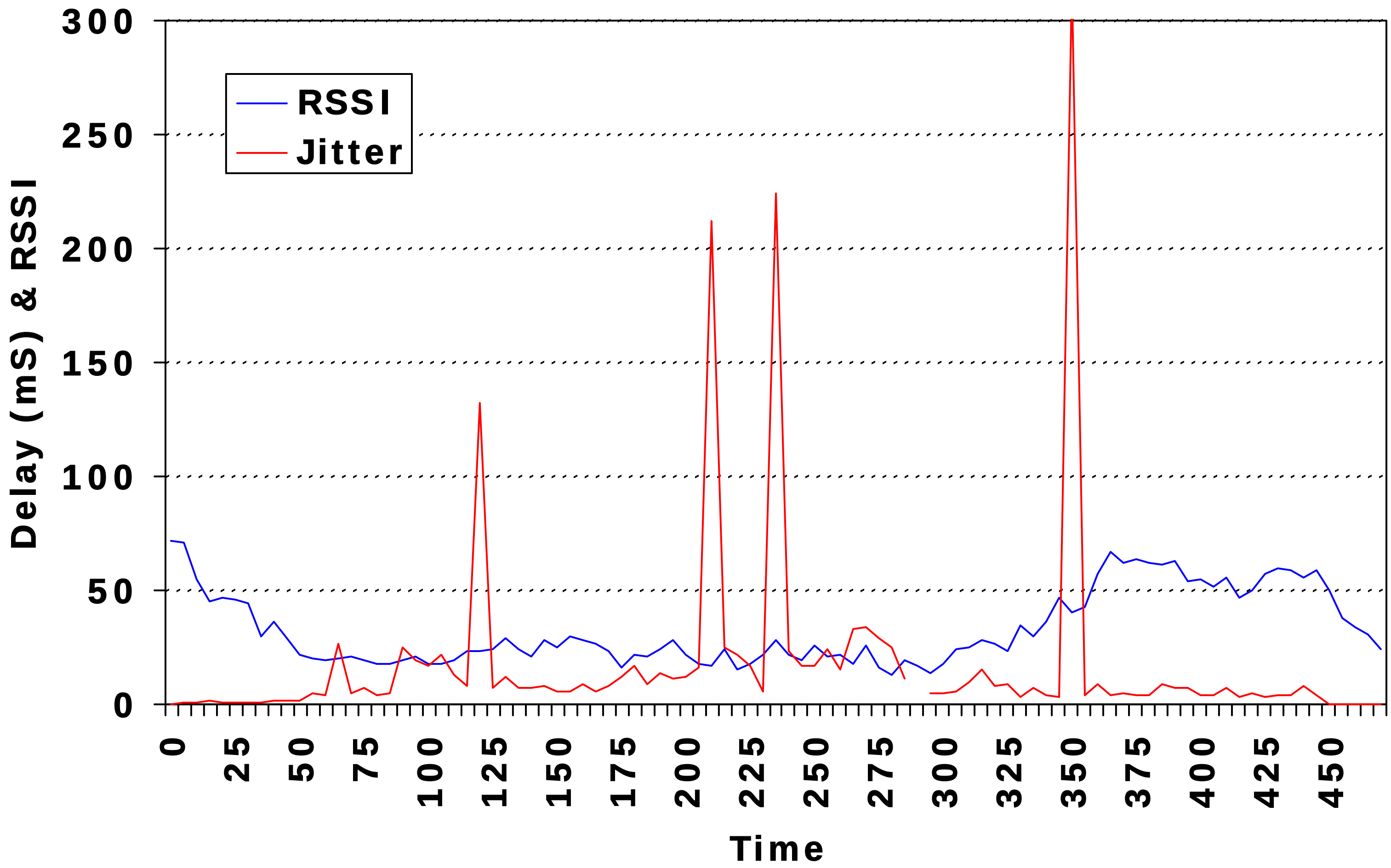


Temporal Clipping occurs with VAD or Echo Suppressors -- gaps in speech, start/end of words missing

Issues related to Wireless

- Handoffs between access points
  - Short gaps as call is handed from one access point to another
- Jitter due to retransmissions
  - 802.11 uses retransmission to improve reliability
  - For low signal strength - leads to increased jitter
- Delay events during speed changes?

Example - RSSI and Jitter for 802.11b WLAN



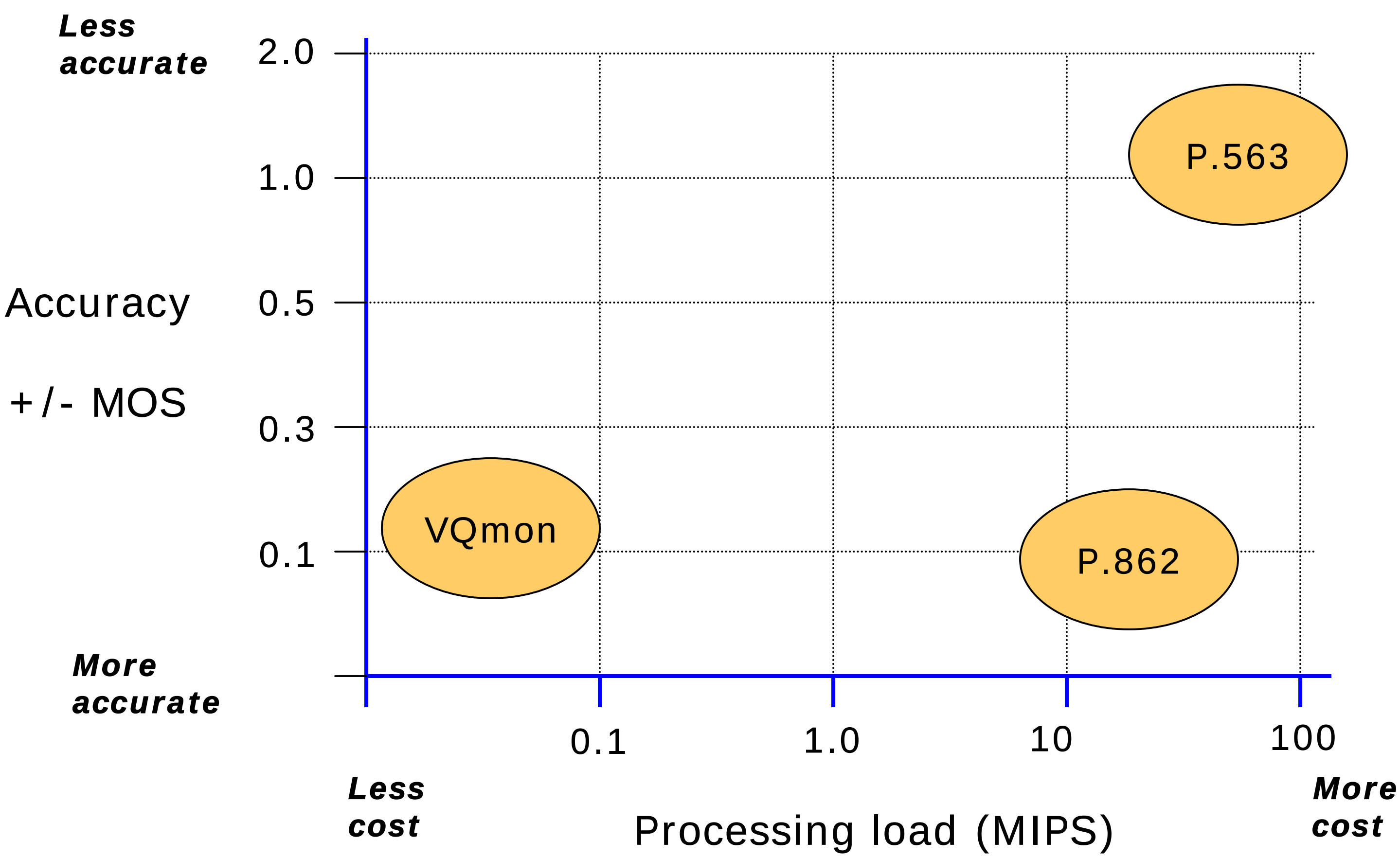
OK - so what do I do about it?

- Tools for measuring VoIP performance
- VoIP QoS reporting protocols and the VoIP performance management architecture
- Equipment requirements
  - Integrated performance monitoring
  - Priority queuing in routers
  - Better jitter buffer and PLC algorithms in endpoints
- Design guidelines

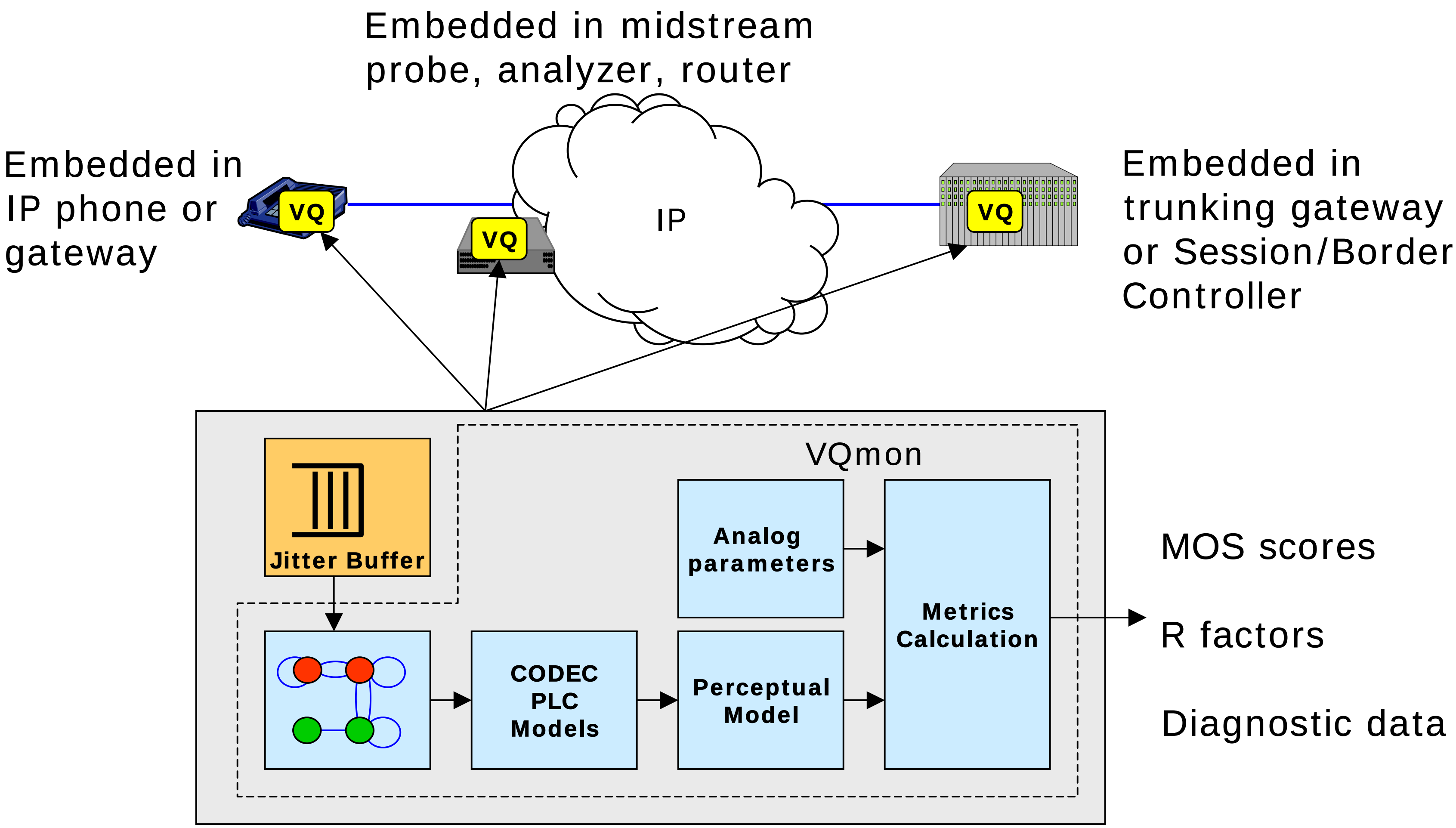
Tools for measuring VoIP performance

	VoIP Specific	Analog signal based
Active Test - Measure test calls	VQmon ITU G.107	ITU P.862 (PESQ)
Passive Test - Measure live calls	VQmon *ITU P.VTQ*	ITU P.563

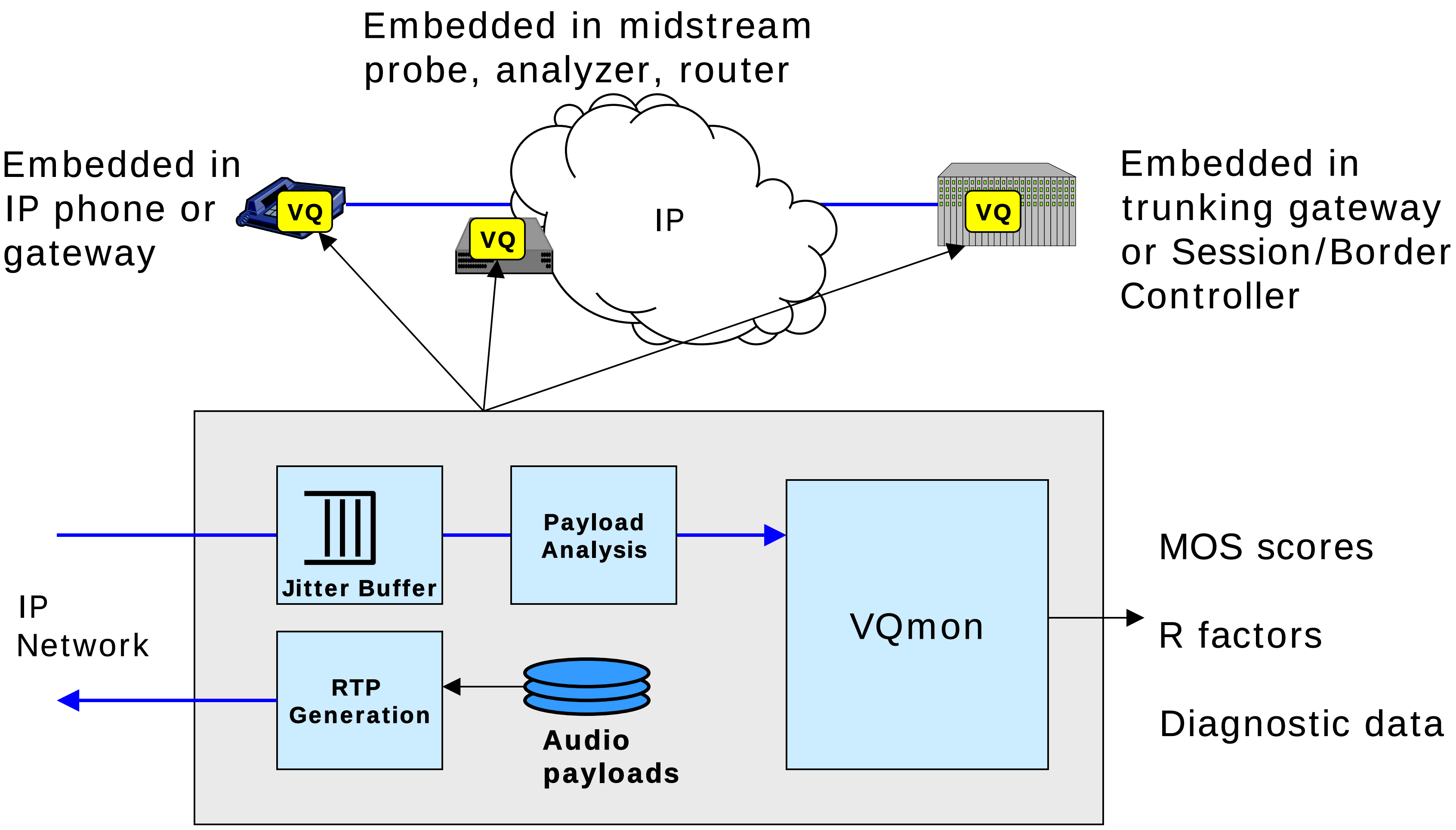
Accuracy and Processing Time comparison



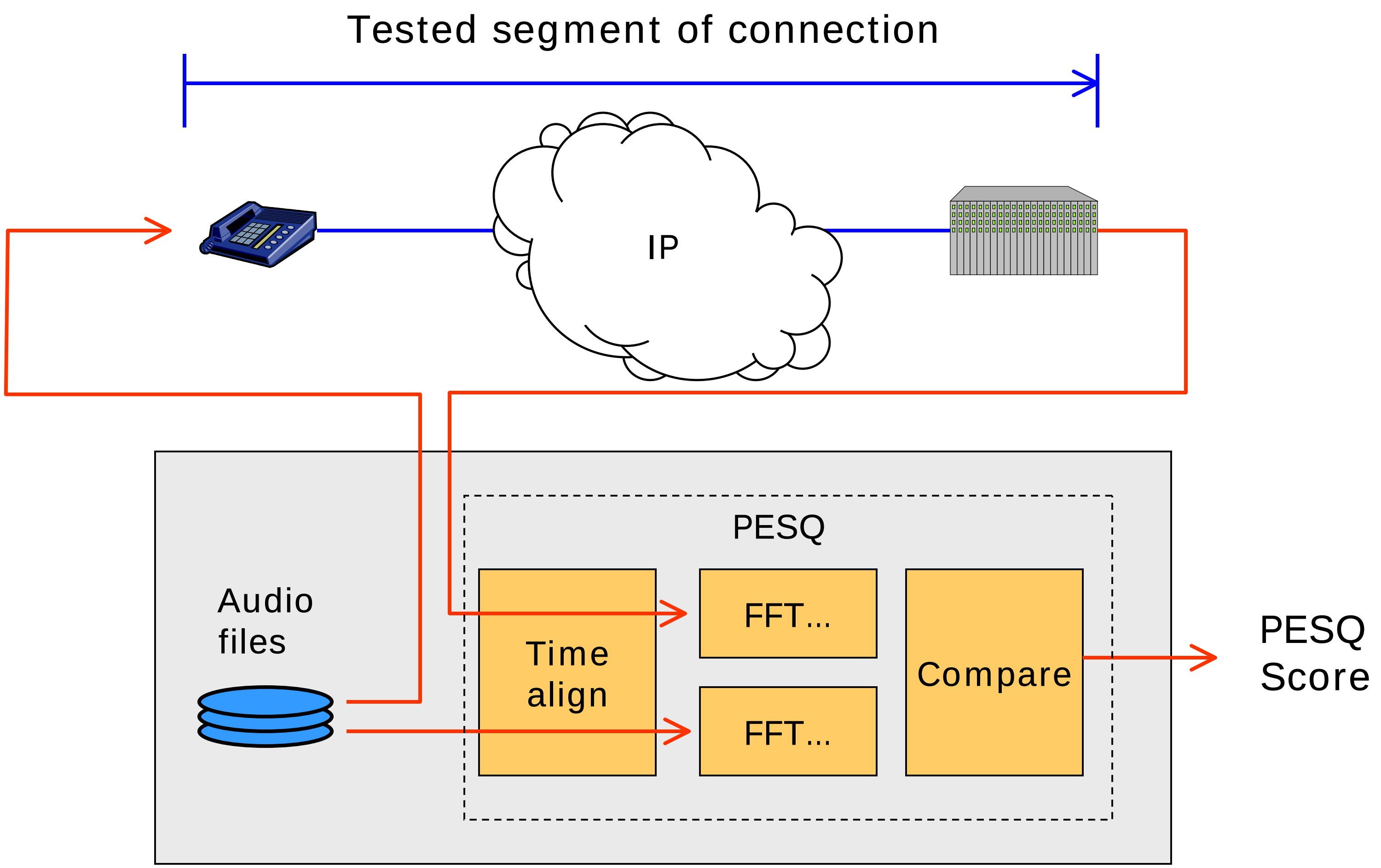
VQmon - passive monitoring application



VQmon - active test application



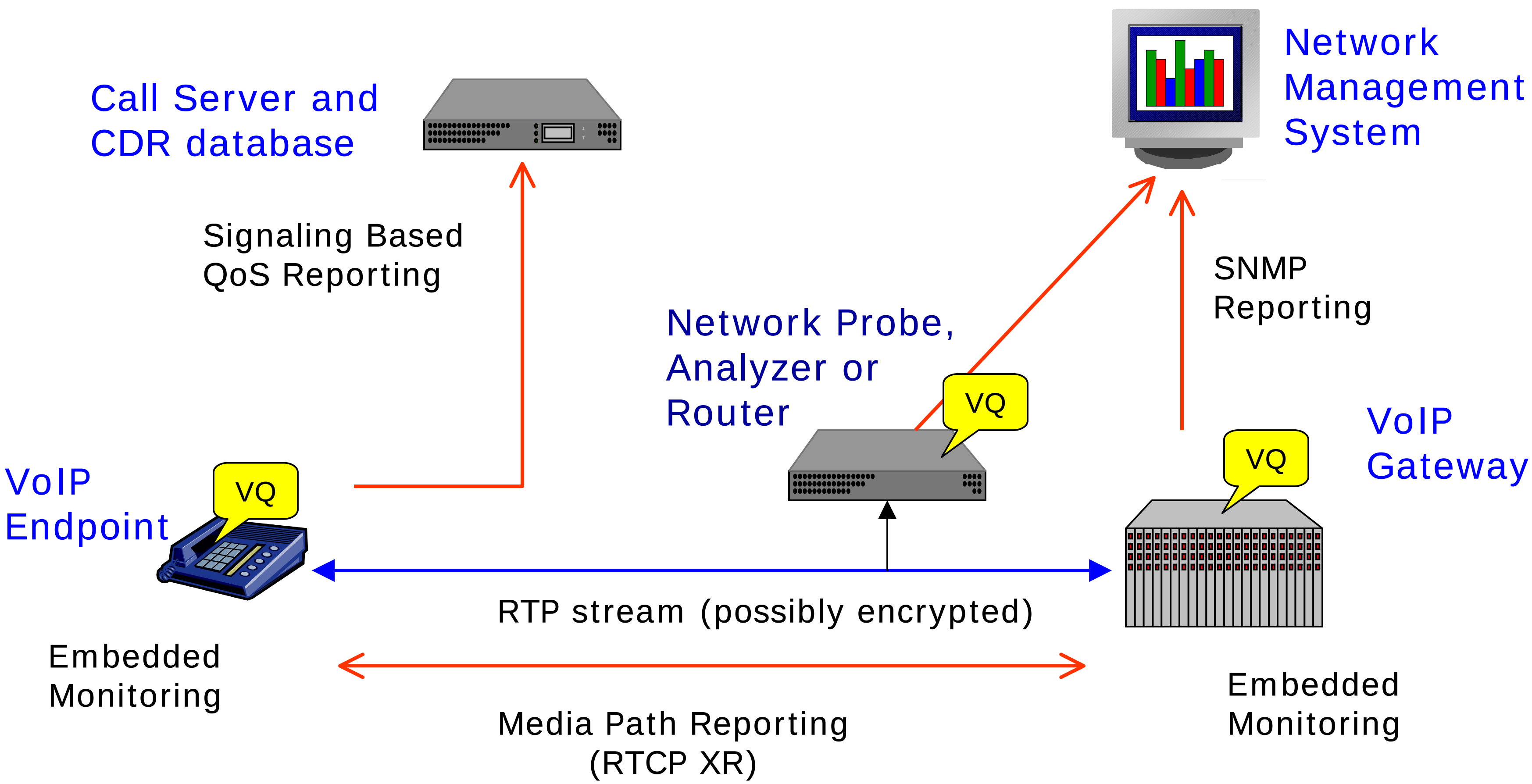
PESQ - active test application



Active or Passive Testing?

- Active testing
  - works for pre-deployment testing and on-demand troubleshooting
- But!!!!
  - IP problems are transient
- Passive monitoring
  - Monitors <u>every</u> call made
  - Captures information on transient problems
  - Provides data for post-analysis
- Therefore - you need both

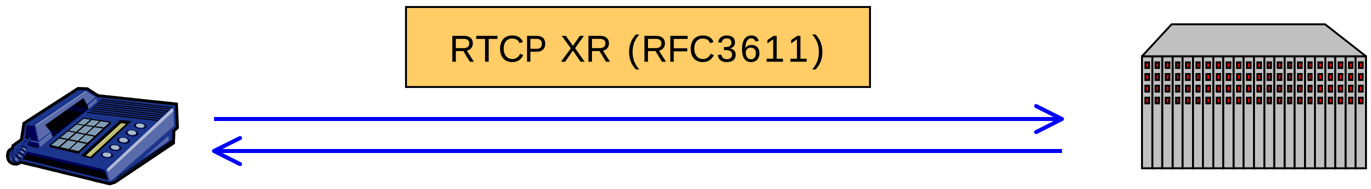
VoIP Performance Management Framework



VoIP Performance Management Framework

- Embedded monitoring function in IP phones, residential gateways....
  - Close to the user
  - Least cost + widest coverage
- Protocol support developed
  - RTCP XR (RFC3611), SIP, MGCP, H.323, Megaco
  - Draft SNMP MIB
- Works in encrypted environments
- Already being deployed by equipment vendors

The role of RTCP XR



- Provides a useful set of metrics for VoIP performance monitoring and diagnosis

- Supports both real time monitoring and post-analysis

- Extracts signal level, noise level and echo level from DSP software in the endpoint

- Exchanges info on endpoint delay and echo to allow remote endpoint to assess echo impact

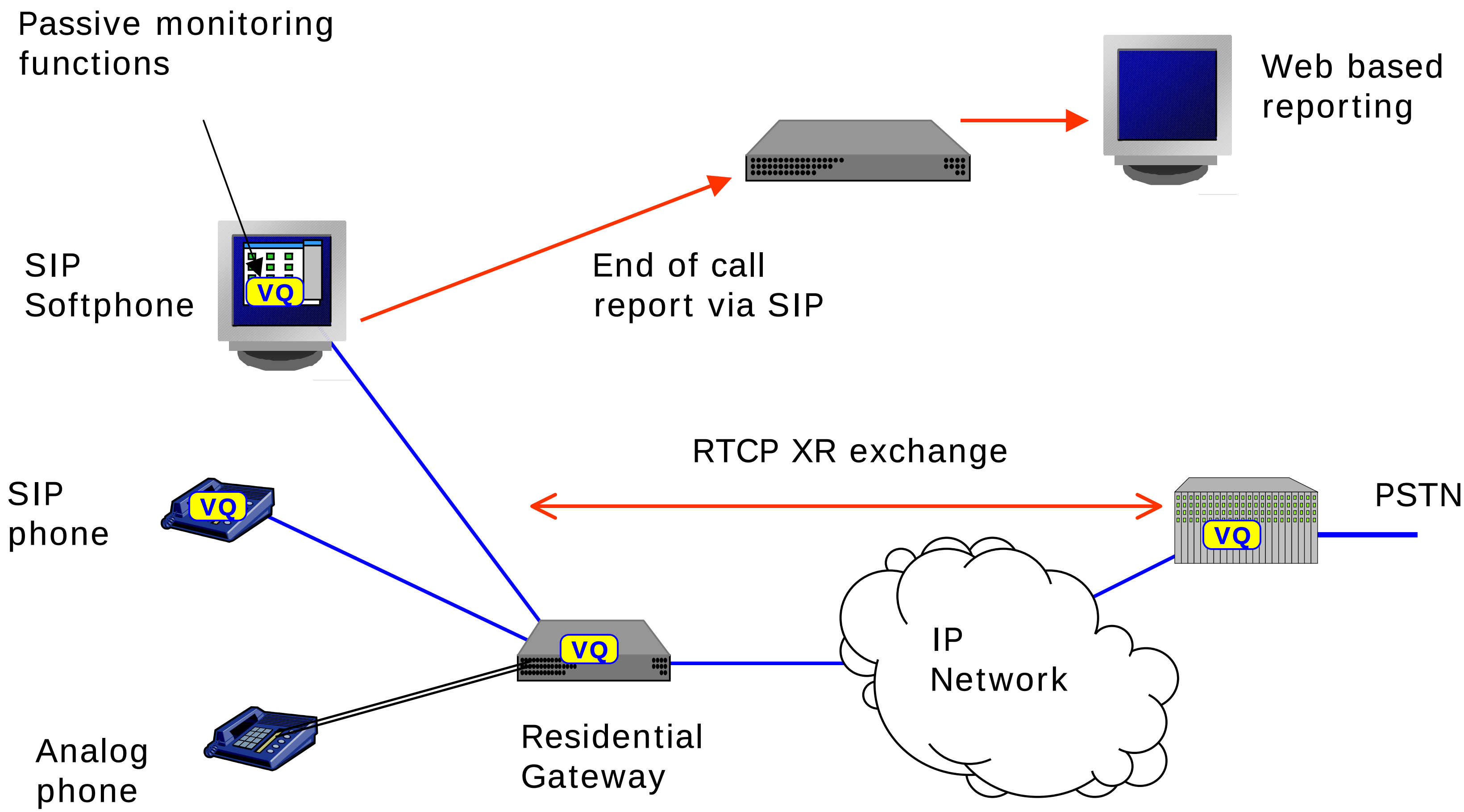
- Provides midstream probes/ analyzers access to analog metrics if secure RTP is used

- Goes through firewalls.........

RTCP XR - RFC3611 - VoIP Metrics block

Loss Rate	Discard Rate	Burst Density	Gap Density
Burst Duration (mS)		Gap Duration (mS)	
Round Trip Delay (mS)		End System Delay (mS)	
Signal level	RERL	Noise Level	Gmin
R Factor	Ext R	MOS-LQ	MOS-CQ
Rx Config	-	Jitter Buffer Nominal	
Jitter Buffer Max		Jitter Buffer Abs Max	

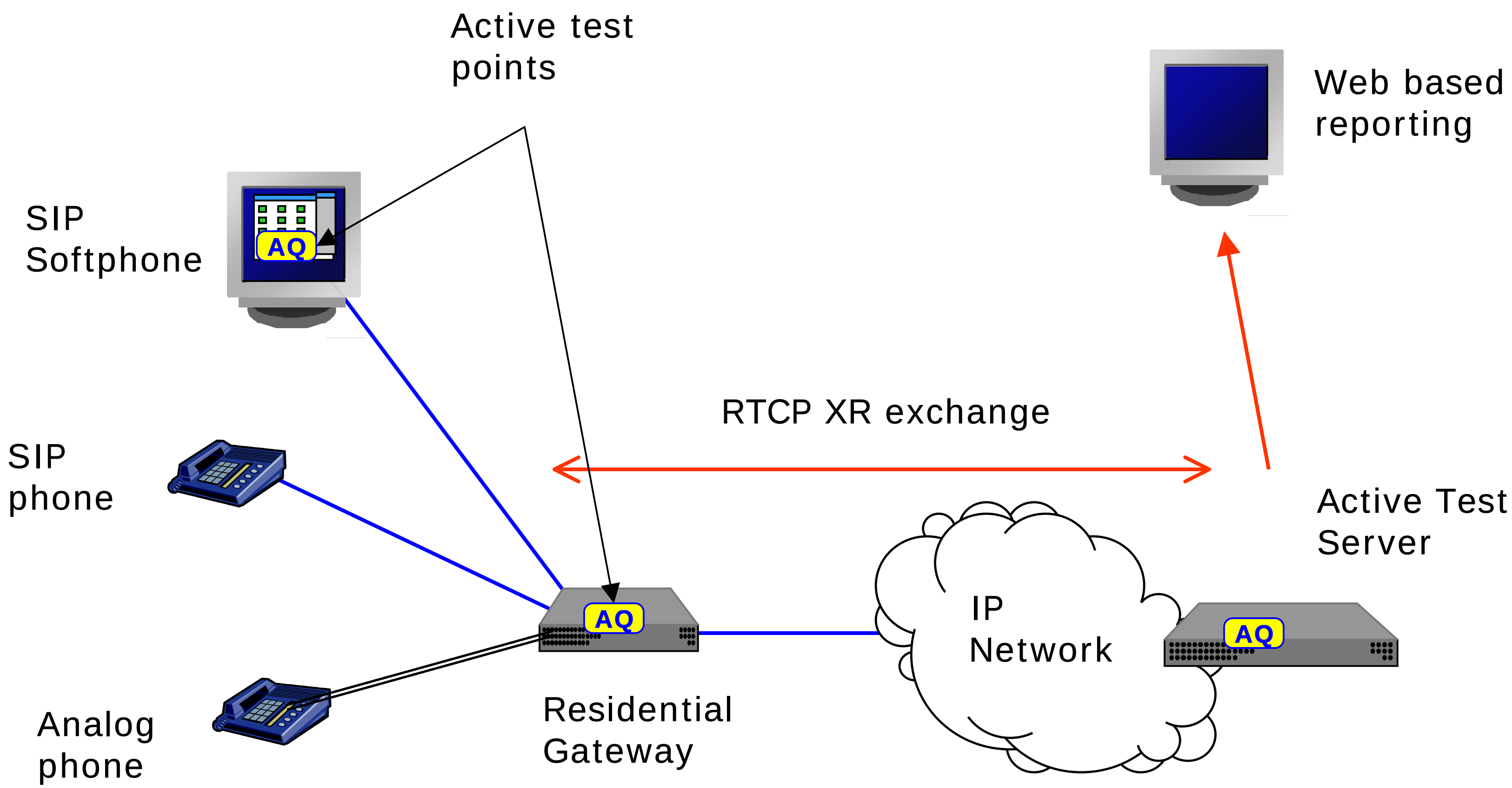
Residential VoIP service application - passive



Applying RTCP XR Metrics

- Discard occurs in 1-2 second bursts
  - Typically access link congestion
- Signal level > -10dBm0
  - Too loud, volume problem in handset or voice port in gateway, could cause clipping
- Signal level < -30dBm0
  - Too quiet, gaps in speech
- Noise level > -55dBm0
  - Noisy signal (background or equipment?)
- RERL < 55dBm and Delay > 50mS
  - Echo problem
- Delay > 300mS (End system + network)
  - Conversational difficulty

Residential VoIP service application - active



Minimizing / Mitigating problems

- Careful network design
- Improved PLC algorithms and robust codecs
- Prioritizing voice traffic
- Call admission control

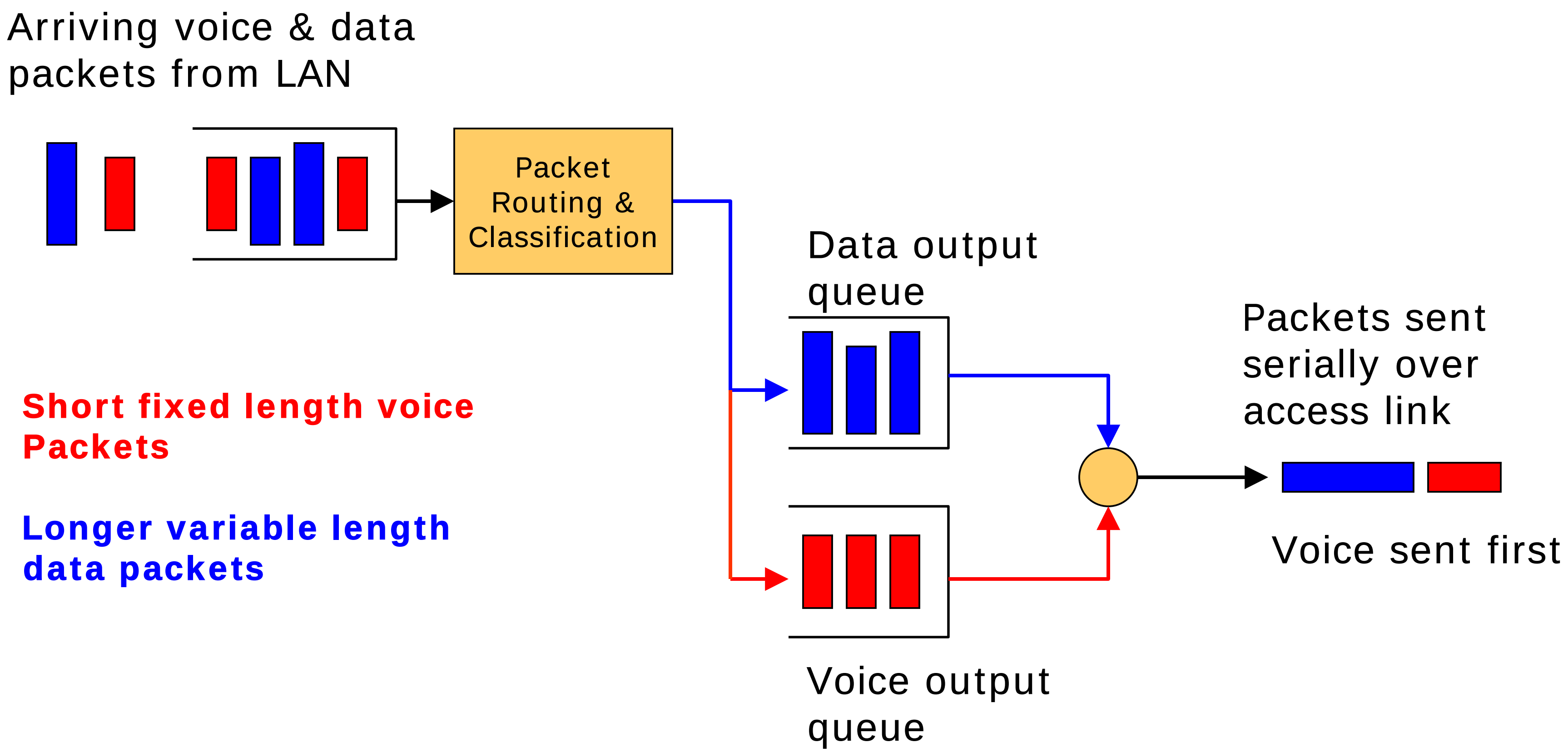
Network design

- Many problems can be easily predicted
  - Enough bandwidth?
  - Too many interfering data sources?
  - No QoS controls?
- Useful tools
  - Application notes (e.g. Telchemy's "Six Steps")
  - Simulation tools
  - Predeployment testing tools

Technologies for improving QoS

- Priority queues for Voice traffic
- Admission control
- Improved Codec/PLC algorithms

Priority queuing in routers



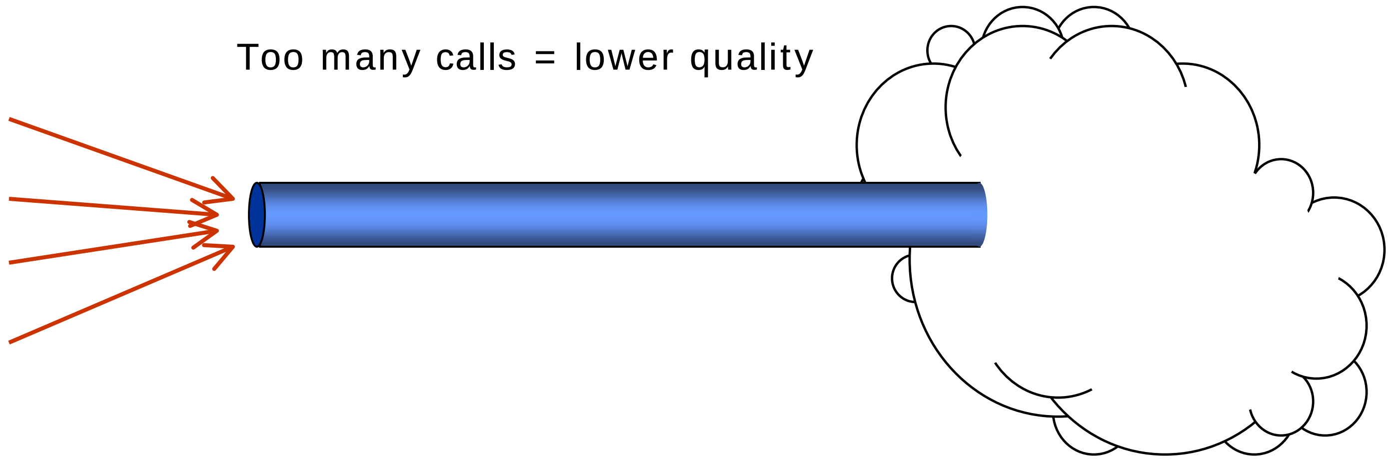
Short fixed length voice Packets

Longer variable length data packets

Priority queuing in routers

- Can get internal congestion in router
  - Packets have to be inspected, NAT'd, moved between internal queues….
- Voice packets still have to wait for one complete data packet
  - May need to also limit MTU size on slower links

Admission control



Call Admission Control
- Limit number of active calls

Measurement based Call Admission Control
- Measure quality and adjust call volume to maintain

Practical Problems
- Codec type can change dynamically (e.g. 8k G729 -> 64k G711
- Applying to complex routes with many bottlenecks

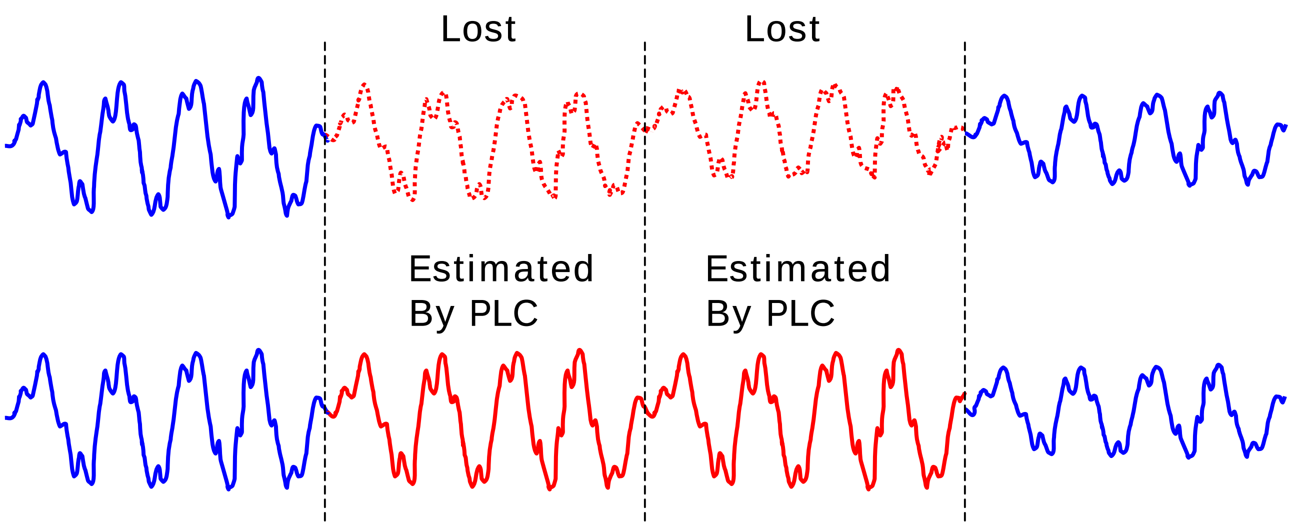
Robust Codecs and PLC Algorithms

- "Good" PLC algorithm
  - Good for isolated lost packets
  - Some distortion for loss rates of 10%
  - Intelligible for loss rates of 20%

- Simple PLC algorithms
  - OK for isolated lost packets
  - Some distortion for loss rates of 5%
  - Intelligible for loss rates of 10%
  - Audible artifacts - beeps, robotic sounds

- BUT
  - Loss is BURSTY therefore PLC algorithms USUALLY have to cope with 20-30% loss rates!!!

Robust Codec and PLC Algorithms



Approaches

- Use Codec that does not depend on previous frame (e.g. ILBC)
- Smarter PLC algorithms that match waveforms, avoid glitches

Planning

- Understand what affects performance
  - This presentation, www.voiptroubleshooter.com
- Understand what your user scenarios are
  - DSL, Cable Modem, Dialup….
- Understand what your users' expectations are…..
  - Cost vs performance
- Understand what equipment would be used
  - Residential gateways - codec types….
- Understand that you can't easily add management as an afterthought - buy/ recommend equipment that supports RTCP XR (properly)
- Develop a management/ troubleshooting strategy, identify tools and technologies

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