Signal Processing for Packet Voice Telephony

IEEE OEB ComSoc, San Ramon
May 20, 2004

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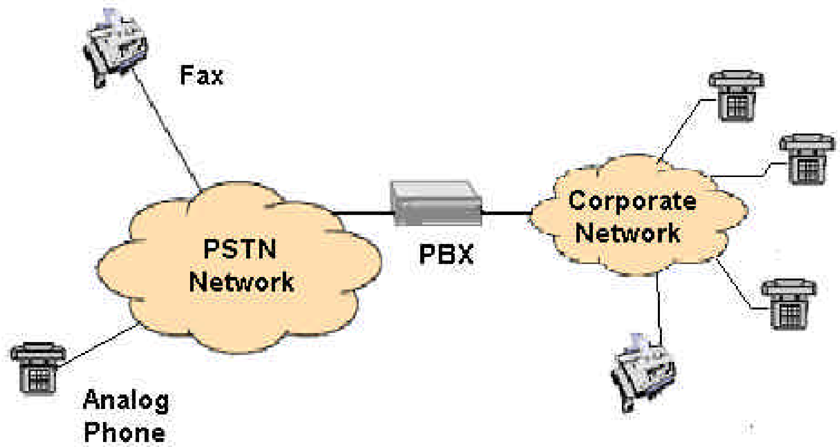
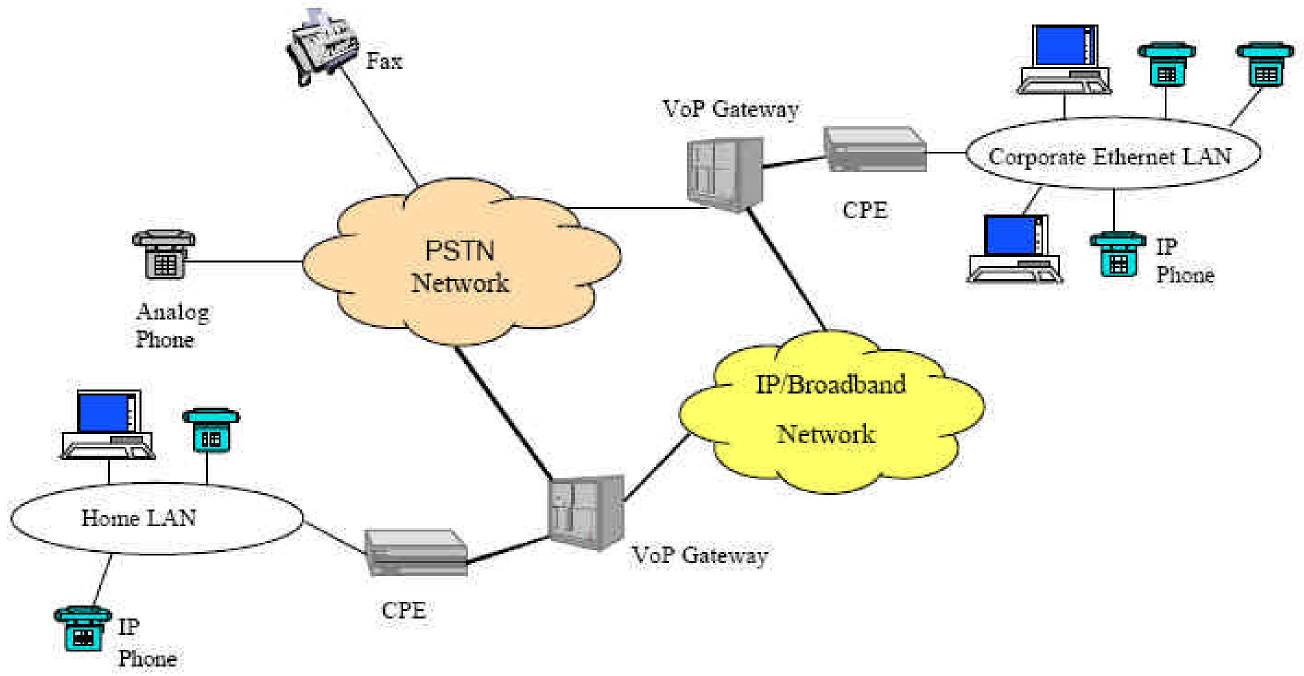
Agenda

- About LSI Logic, briefly ...
- Packet Voice Networks Overview
- Signal Processing Layer Internals
  - Voice-specific functions
    - Speech codecs; packet loss concealment; echo cancellation; silence suppression; …
  - Other related functions
- Processing needs for different contexts
- A few trends & issues

Voice Networks: Past



Voice Networks: Present



Telling Quotes

"If you don't do it, next year or the one after that you won't be playing in the game"

Richard Notebaert (CEO, Qwest)

(while announcing Qwest's plan to offer low-cost IP based phone service, 2003).

Telling Quotes

"As of three weeks ago, all the long-distance (voice) traffic in Italy is carried over the IP network"

Stefano Pileri (Head, Domestic Network, Telecom Italia)

(announcement made at ITU Telecom Conference, Geneva October 15, 2003).

Telling Quotes

"Telecom may be heading the way of DRAMs, where the price is set by the most idiotic competitor. ... It is a race to the bottom, and the bottom in this case is free service"

Robert Lucky

(Keynote Speech, Communications Design Conference, San Francisco, March 31, 2004).

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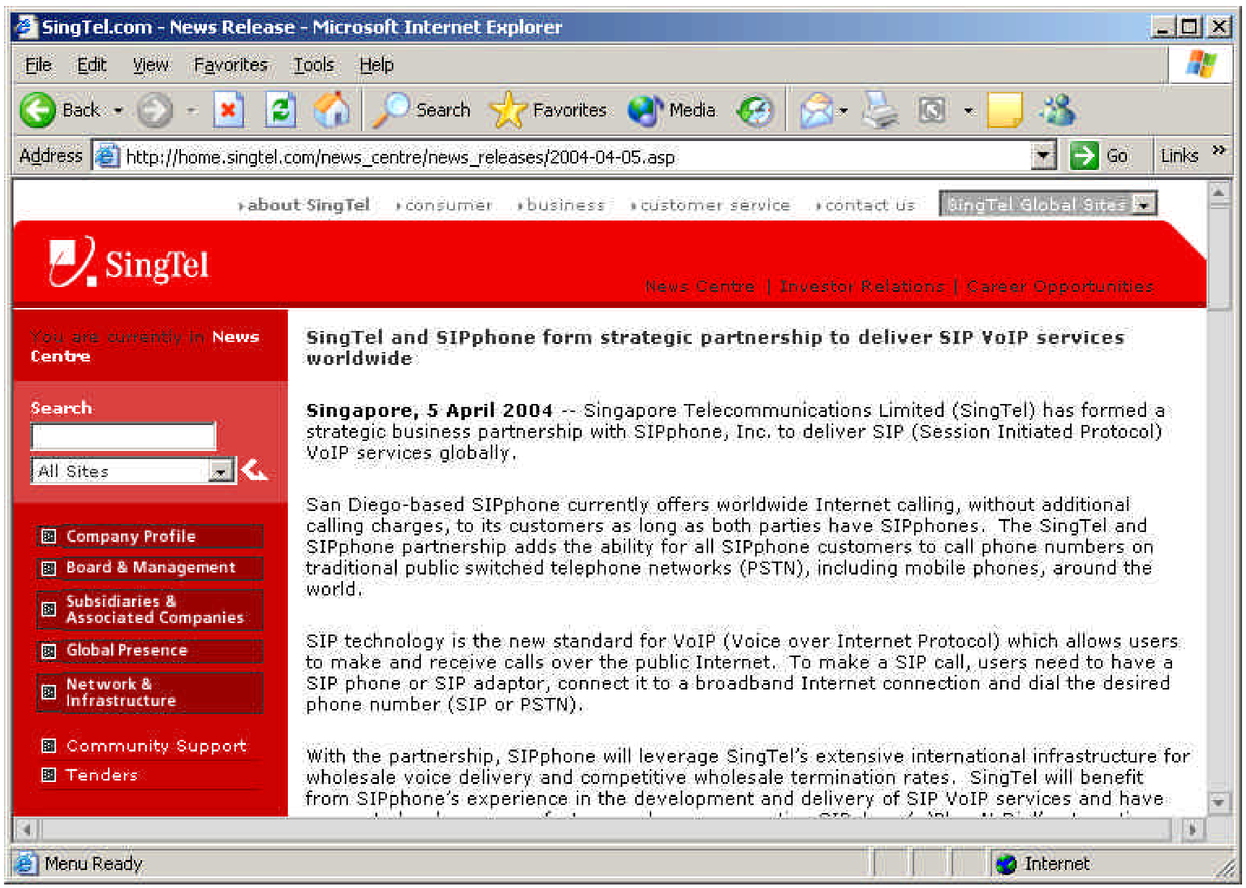
Not so SIPle News

Singtel + SIPphone Press Release (April 5, 2004)
(home.singtel.com/news_centre/news_releases/2004-04-05.asp)



SingTel and SIPphone form strategic partnership to deliver SIP VoIP services worldwide

Singapore, 5 April 2004 -- Singapore Telecommunications Limited (SingTel) has formed a strategic business partnership with SIPphone, Inc. to deliver SIP (Session Initiated Protocol) VoIP services globally.

San Diego-based SIPphone currently offers worldwide Internet calling, without additional calling charges, to its customers as long as both parties have SIPphones. The SingTel and SIPphone partnership adds the ability for all SIPphone customers to call phone numbers on traditional public switched telephone networks (PSTN), including mobile phones, around the world.

SIP technology is the new standard for VoIP (Voice over Internet Protocol) which allows users to make and receive calls over the public Internet. To make a SIP call, users need to have a SIP phone or SIP adaptor, connect it to a broadband Internet connection and dial the desired phone number (SIP or PSTN).

With the partnership, SIPphone will leverage SingTel's extensive international infrastructure for wholesale voice delivery and competitive wholesale termination rates. SingTel will benefit from SIPphone's experience in the development and delivery of SIP VoIP services and have

Packet Pathways

- Wired
  - Cable Modems
  - DSL Networks
  - Corporate Ethernet LANS
  - Managed / Public Wide-area Networks (WANs)
- Wireless
  - WLANS (WiFi)
  - Satellite IP Networks (including VSATs)
  - BWA (WiMax) (?)

Protocol Menu

- Protocols for VoIP Inter-operability:
  - MGCP (CableLabs / PacketCable)
  - H.323 (ITU, videoconferencing)
  - H.248 /Megaco (ITU)
  - SIP (IETF)
- Others key protocols (Quality, …):
  - RTP, RTCP
  - UDP
  - RSVP, DiffServ, …

Packet Voice "Boxes"

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VOIP Market: DSP Slice

	2001	2006	
Annual Growth (voice channels):	**12 Million**	**560 Million**	
DSP Revenue:	**US$129M**	**US$1400M**	
ASP (DSP + S/W)	**$25**	**$225**	**Gateway Market Segment**
Channels / DSP	**< 5**	**~ 88**	
Average cost / channel	**$5.70**	**$2.60**	

Source: VOIP & Packet Voice DSP Markets, Forward Concepts, AZ (April 2002)

Packet Voice: Key Hurdles

- Delay
  - Typical end-to-end delays around 100-200ms
- Packet Jitter
  - Typical arrival time jitter around 20-50ms
- Packet Loss
  - Typical losses around 1-2%

Delay: G.114 Guidelines

One-way Delay	ITU-T Classification (with echo "adequately controlled")
< 150ms	Mostly acceptable.
150-400ms	Acceptable (maybe).
> 400ms	Unacceptable (in general).

TYPICAL DELAYS

Terrestrial, national long distance PSTN: < 50ms
Terrestrial, international PSTN: ~ 100ms
Cellular: Mobile to PSTN: ~ 150ms
Cellular: Mobile to Mobile: ~ 300 – 400ms

Delay in Packet Networks

Overall delay break-up:

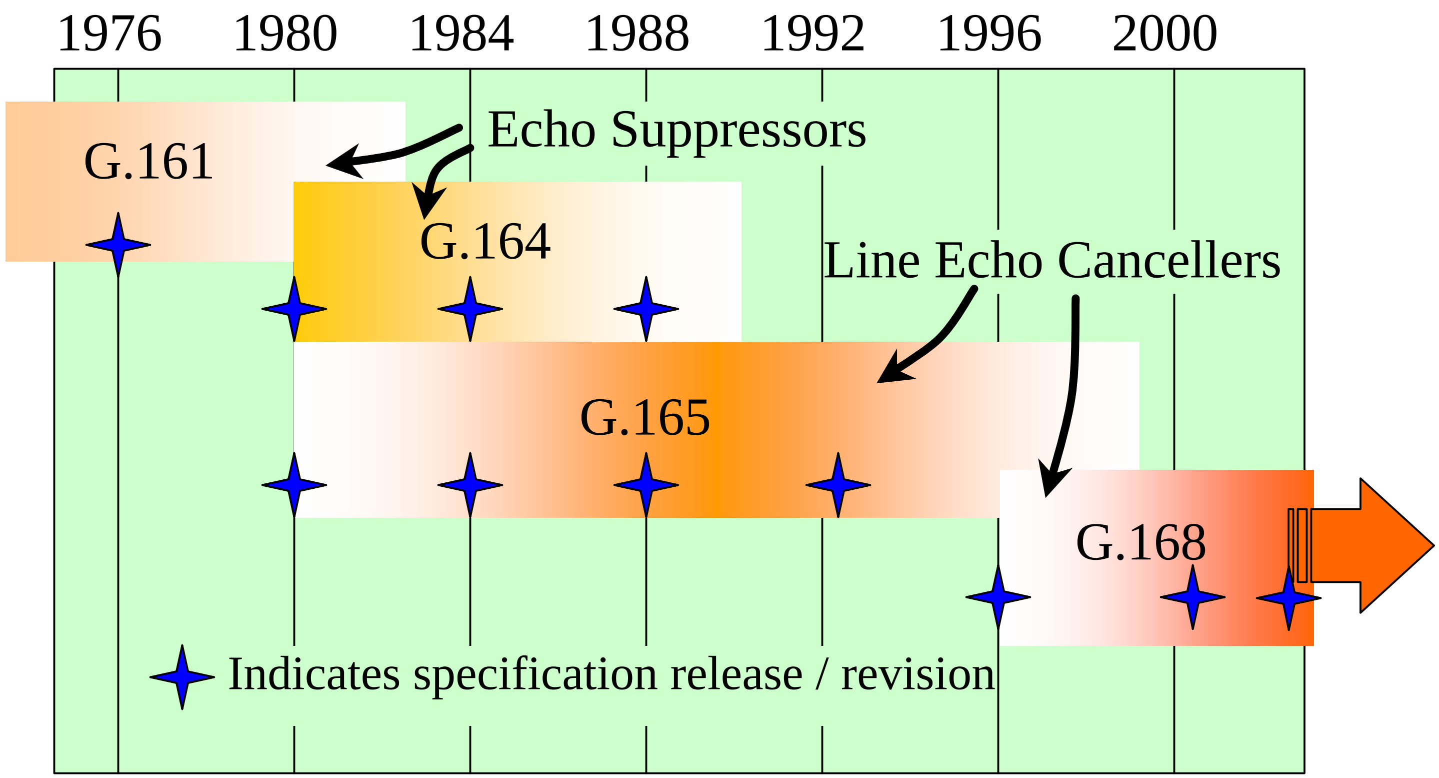
Speech Codec	0.2 - 68ms	Low delay for PCM, ADPCM, G.728, …
Packetization	5 - 30ms	
Interleaving	0 - 60ms	Optional
Transmission	25 - 150ms	
Jitter Buffer	50 - 100ms	
Total	~80 - 400ms	**Typical: 100 – 200ms**

Revised from "Internet Telephony: Going like crazy", by G. Thomsen, Y. Jani, IEEE Spectrum, May 2000.

Handling Delay (Echo)

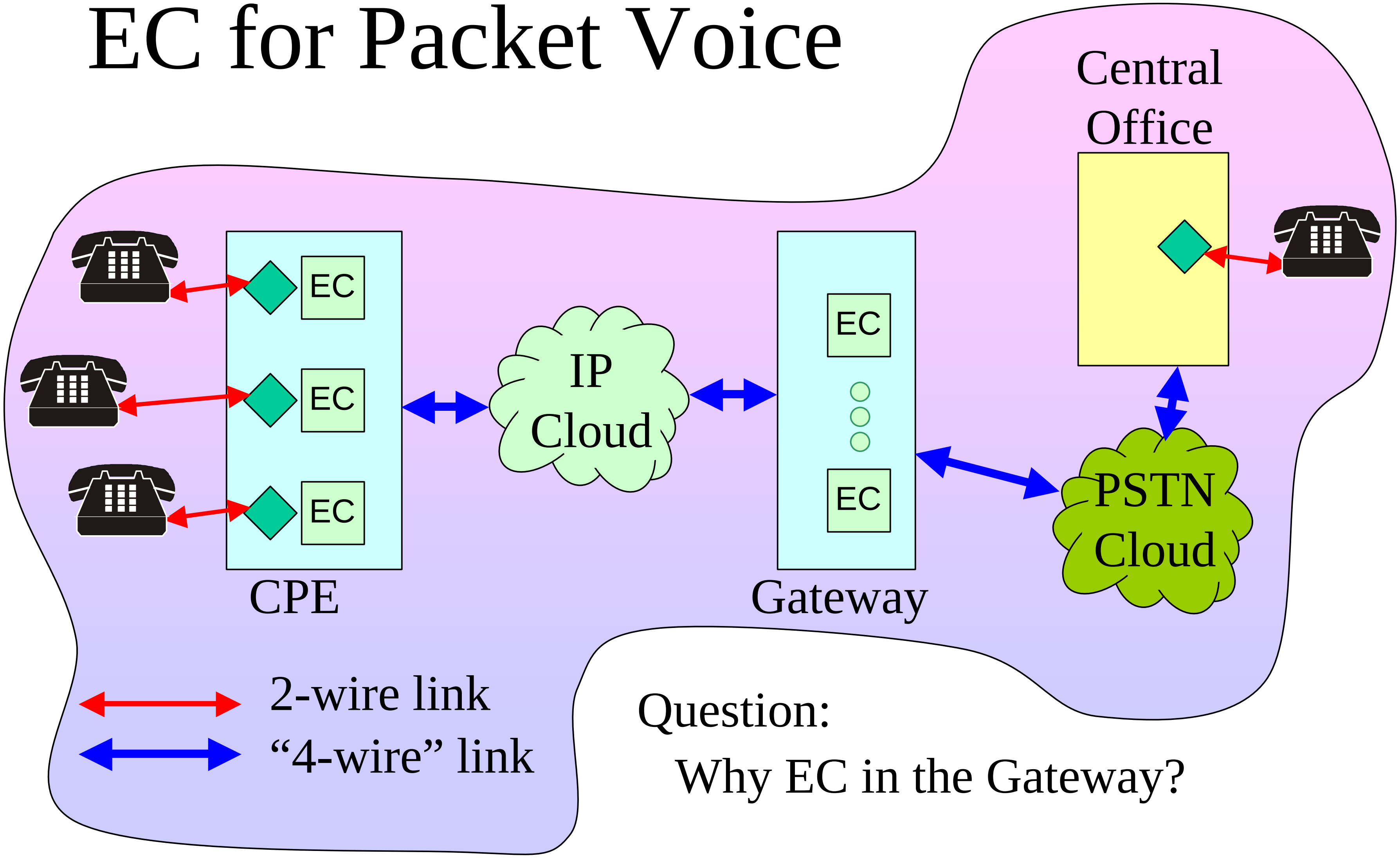
- One-way delays in packet voice networks > 100ms
- As recommended in ITU-T G.131, a network echo canceller (EC) is required.
- EC required only for:
  - PSTN interfaces on voice gateways
  - Analog phone (SLIC) interfaces on CPEs
- EC not required for digital IP phones (AEC is a different option)
- EC tail length – a much misused parameter
- ITU-T G.168 EC was initially developed for PSTN. Can it be applied as-is for packet voice networks?

Tackling Echo: ITU Standards



CCITT → ITU

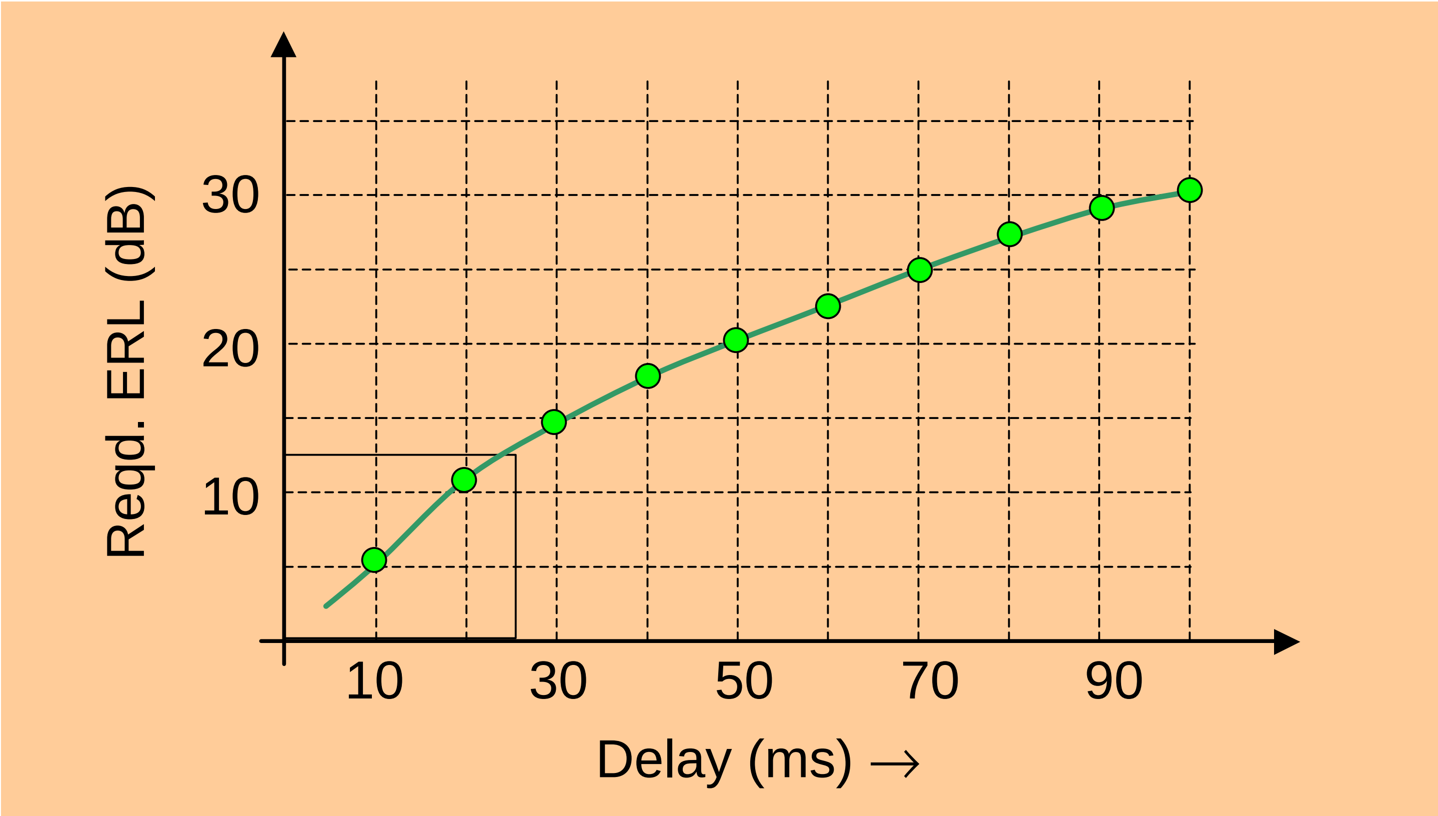
EC for Packet Voice



Question:

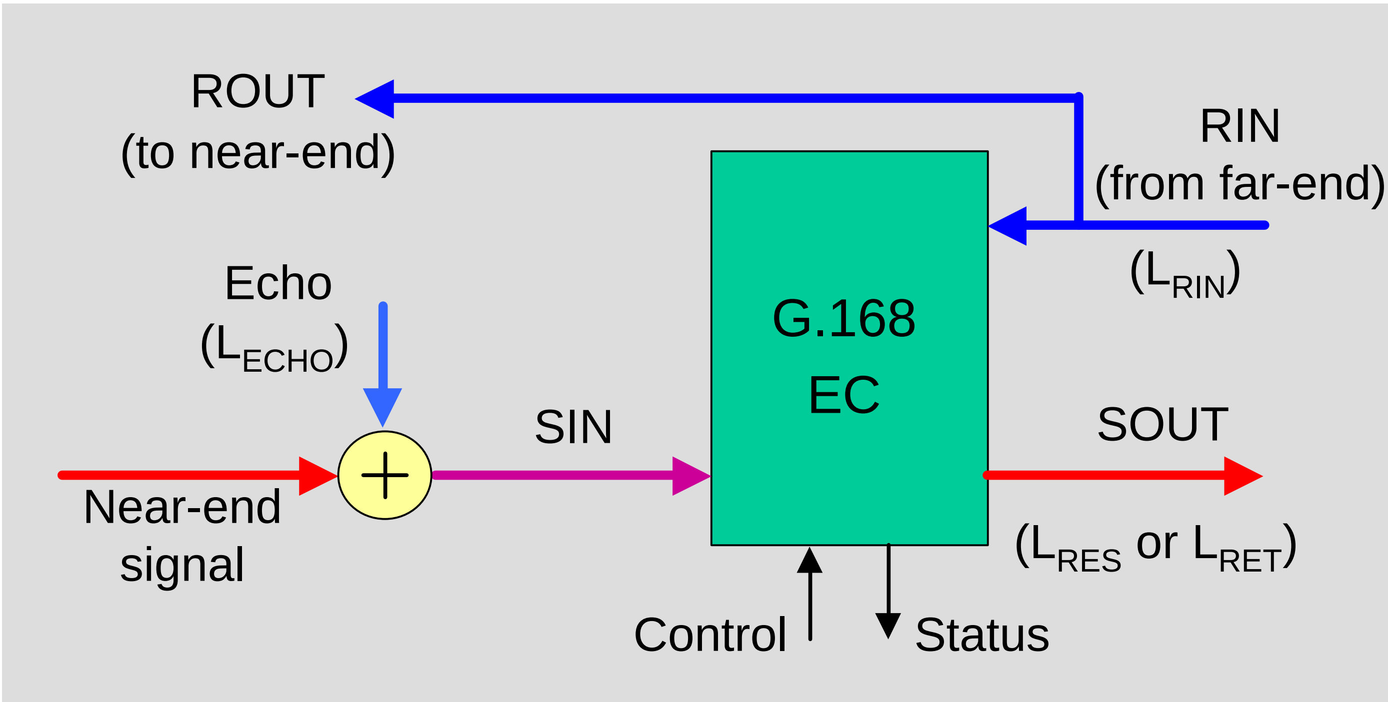
Why EC in the Gateway?

Echo Level and Delay



ERL data from Table 1.1, “Acoustic Signal Processing for Telecommunication”, S. L. Gay and J. Beneste (Ed.s), Kluwer Academic Publishers (2000)

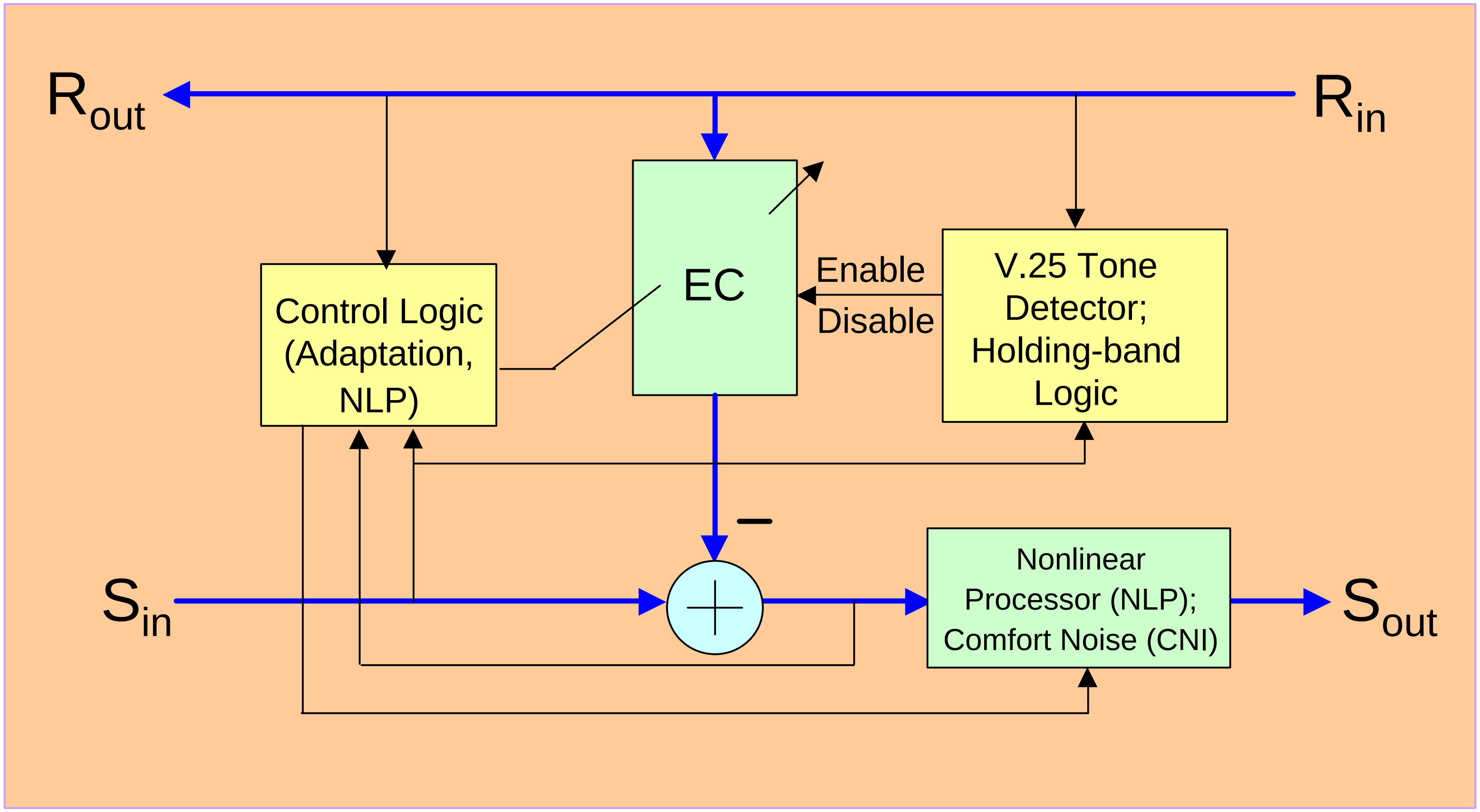
EC – A Black-box View



ERL = L_{RIN} – L_{ECHO}

A_{COM} = L_{RIN} – L_{SOUT} (near-end signal absent)

G.168 EC Internals



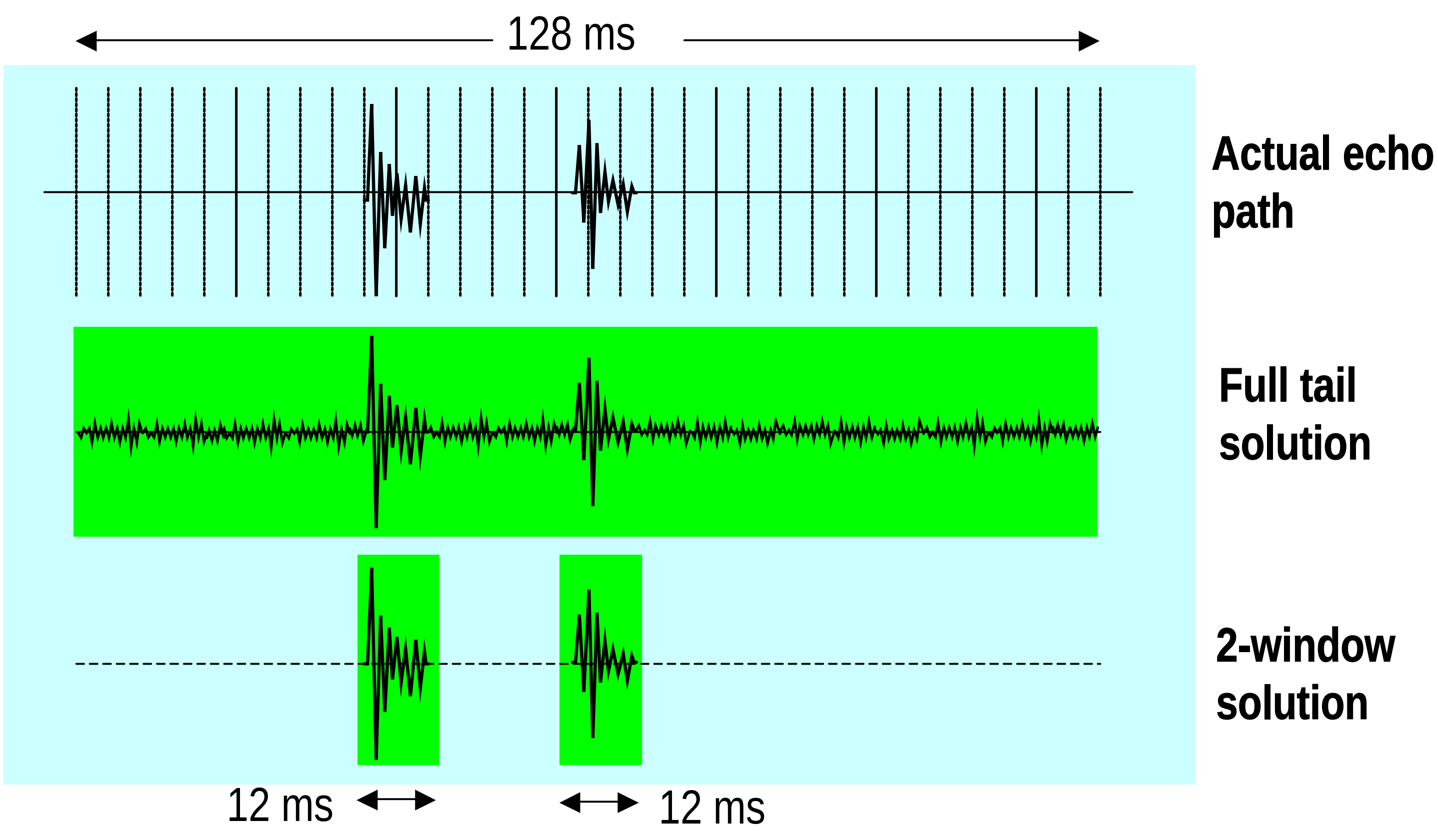
Some EC Design Options

"Full tail" ↔ "Tail independent" or "Floating window"

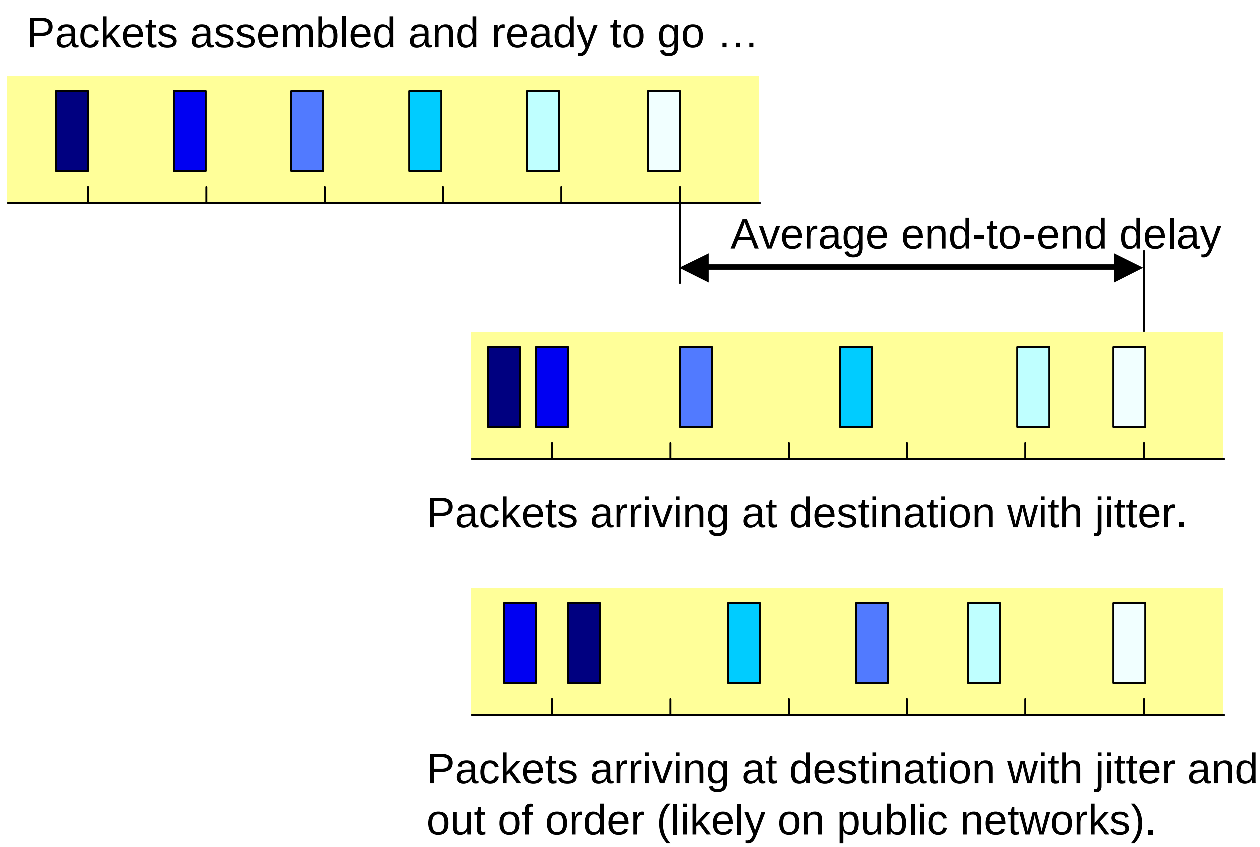
Single filter with robust control ↔ Double filter with simpler controls

Time domain ↔ Transform domain ↔ Subband structure

Full Tail / Floating Window



Handling Packets



Jitter Buffer

- Evidently, jitter buffer is a crucial module in the receiver.
- Out-of-order packet arrivals can be sorted based on RTP time stamp.
- Trade-off of voice quality versus latency.
  - A small buffer helps minimize the extra latency, but drops packets that arrive too late.
- Adaptive jitter buffer that grows or shrinks as needed, is one solution.

Packet Losses

- In addition to packet drops by jitter buffer, packet losses are likely due to
  – UDP (does not offer guarantee of delivery)
  – Network congestion (bandwidth)
  – Router overload (packet throughput)
- Up to 5% (or more?) packet losses considered likely
  – Even 1% packet loss degrades voice quality significantly
  – Packet Loss Concealment (PLC) is yet another essential module in the receiver
  – PLC is not sufficient to handle certain tone signals (DTMF digits, V.25 tone for EC disabling, etc.)

Tone Relay

- Helps in reliable transfer of DTMF digits and other signaling tones (packet losses)
- Fast DTMF detection also avoids possible leakage problems
  - Fast detection particularly important with low bit rate voice codecs such as G.723.1 or G.729.
- Q: Does tone relay use UDP or TCP?

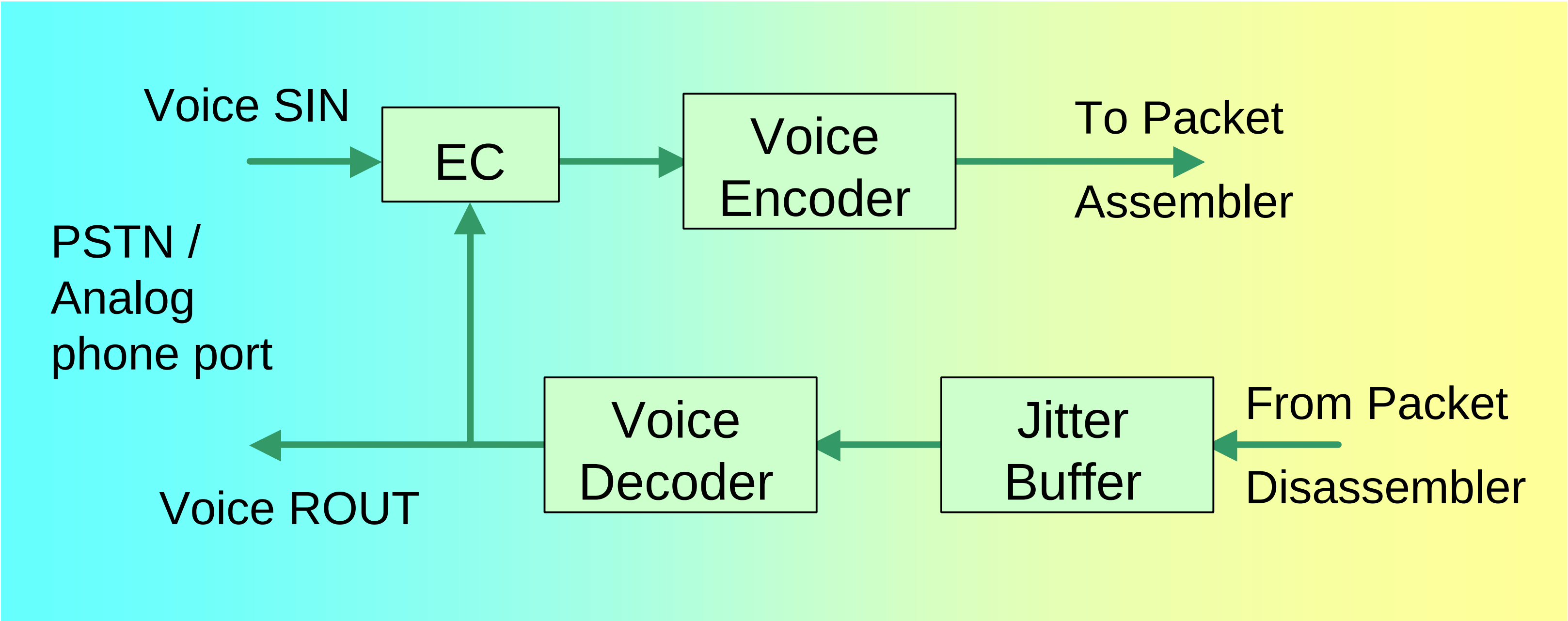
Dealing With Packet Loss

- Network Level (transparent to DSP)
  - QoS protocols
  - Call Admission Control
- Other Non-transparent Means
  - Adaptive Jitter Buffer
  - Interleaving
  - Transmit Redundant Packets
  - Silence Suppression

Quality gained at the cost of extra latency

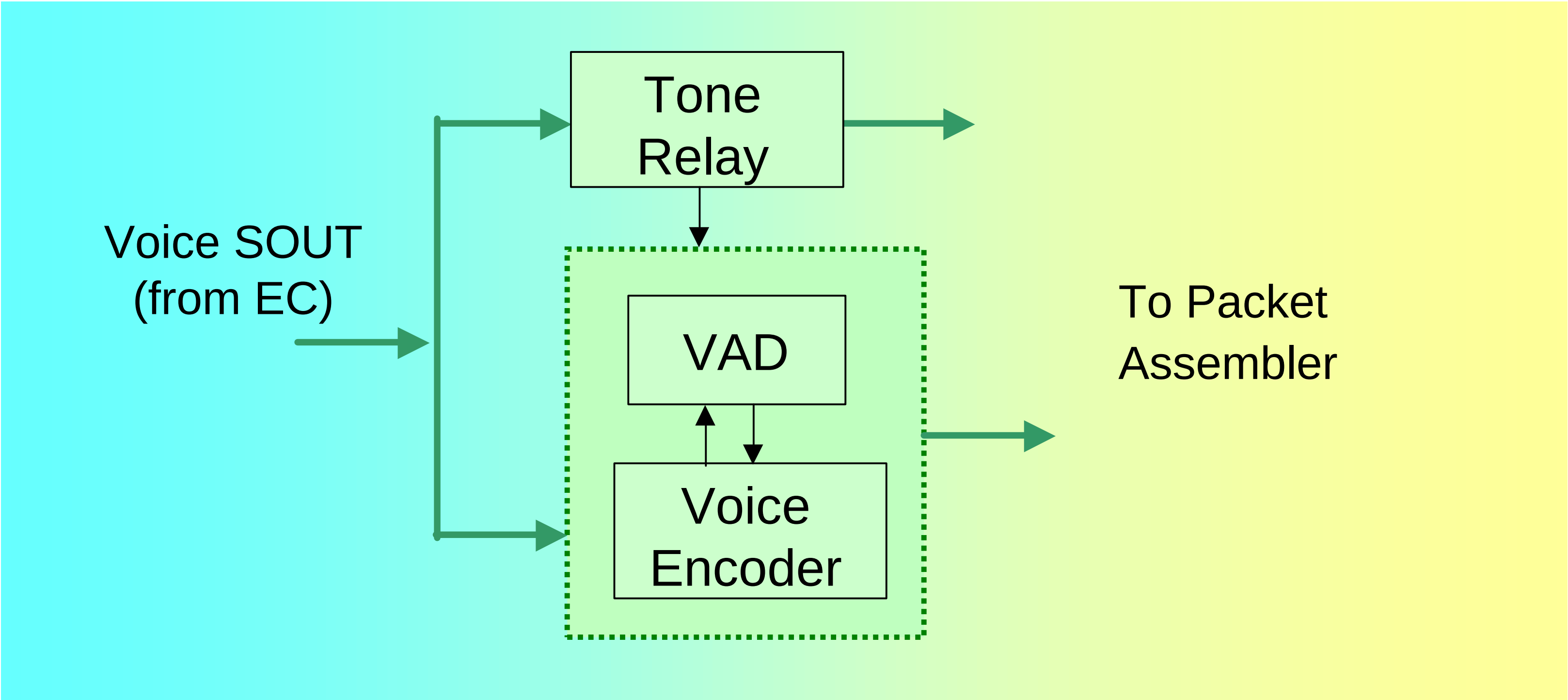
Indirect approach – reduce network congestion

Packet Voice: Key Blocks



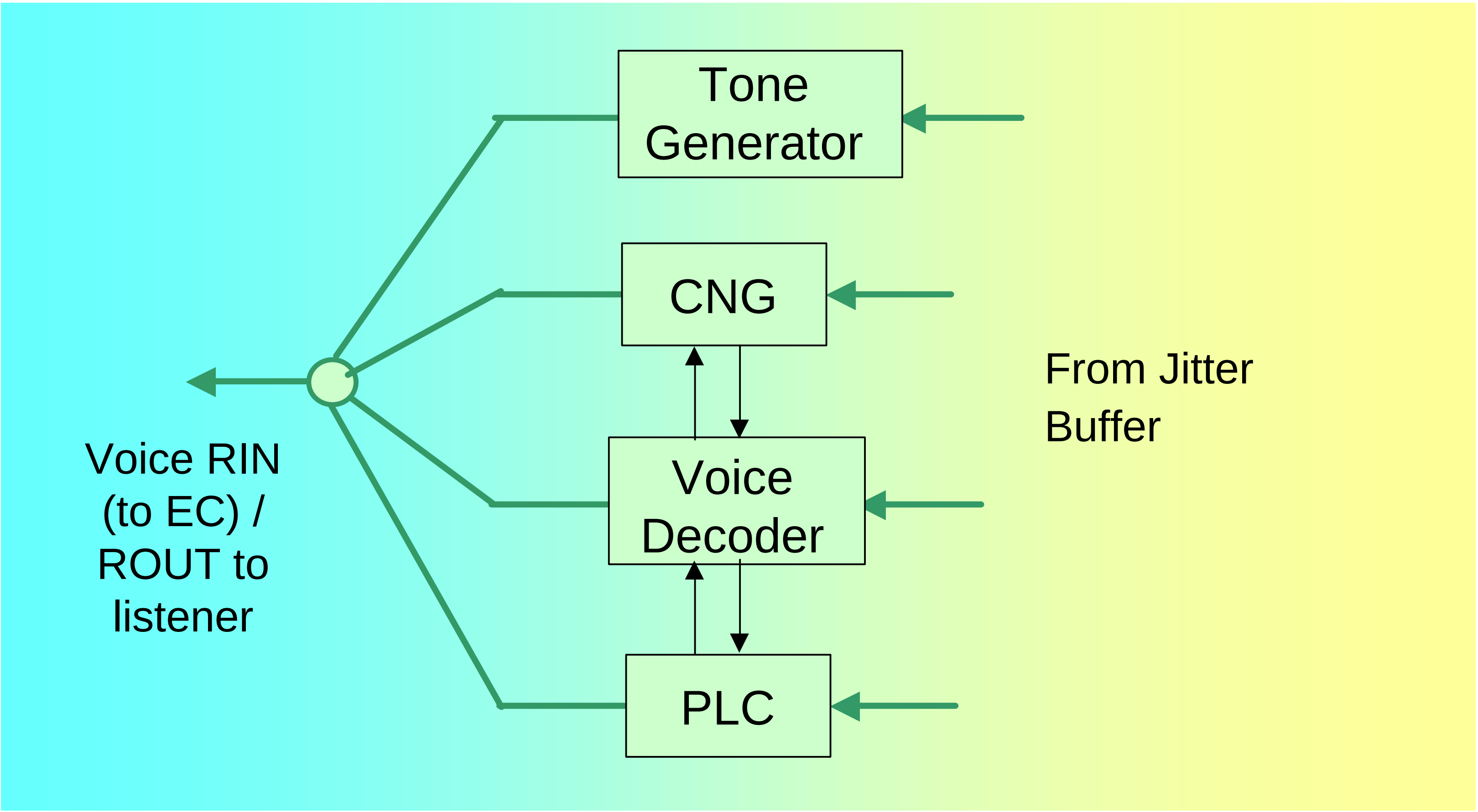
- EC required only if echo can occur in ROUT / SIN loop.
- EC (if deployed) should be properly aligned with any I/O buffering delays in the loop.

Packet Voice: SND Path Detail

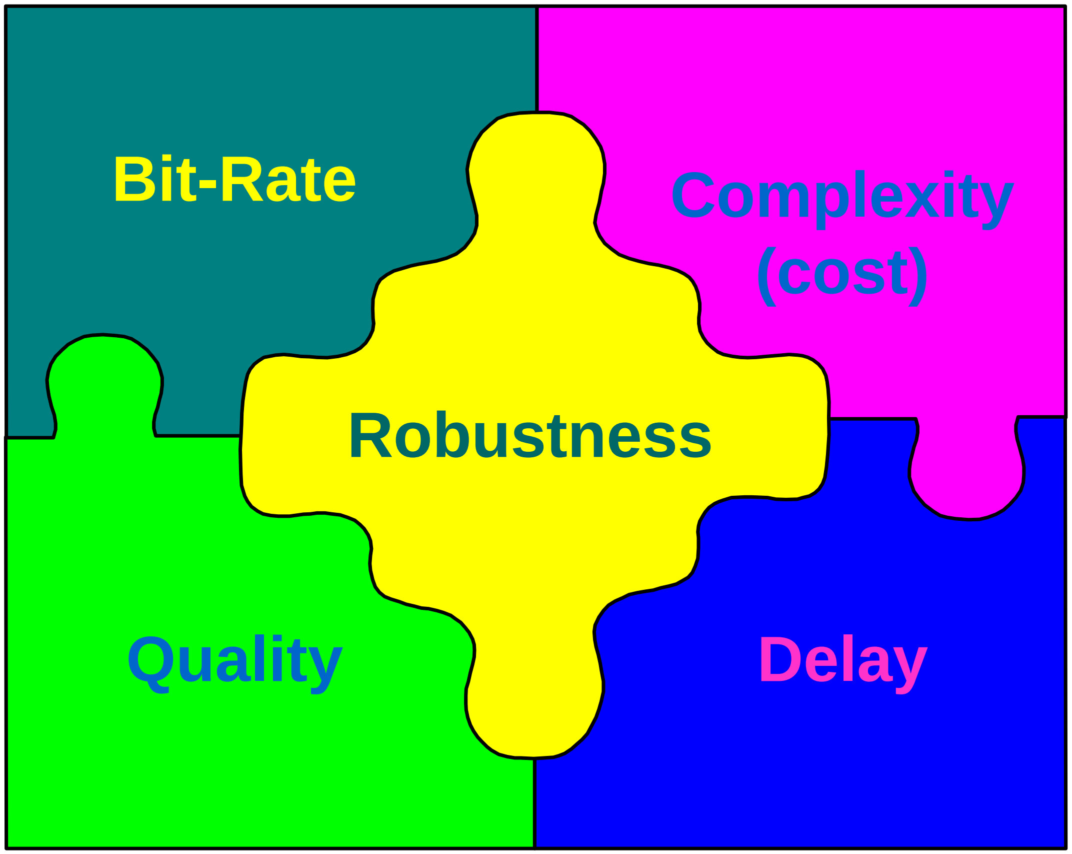


- If an EC is present, how should its NLP + CNI module be used? Preferable to integrate this function with the VAD.

Packet Voice: RCV Path Detail

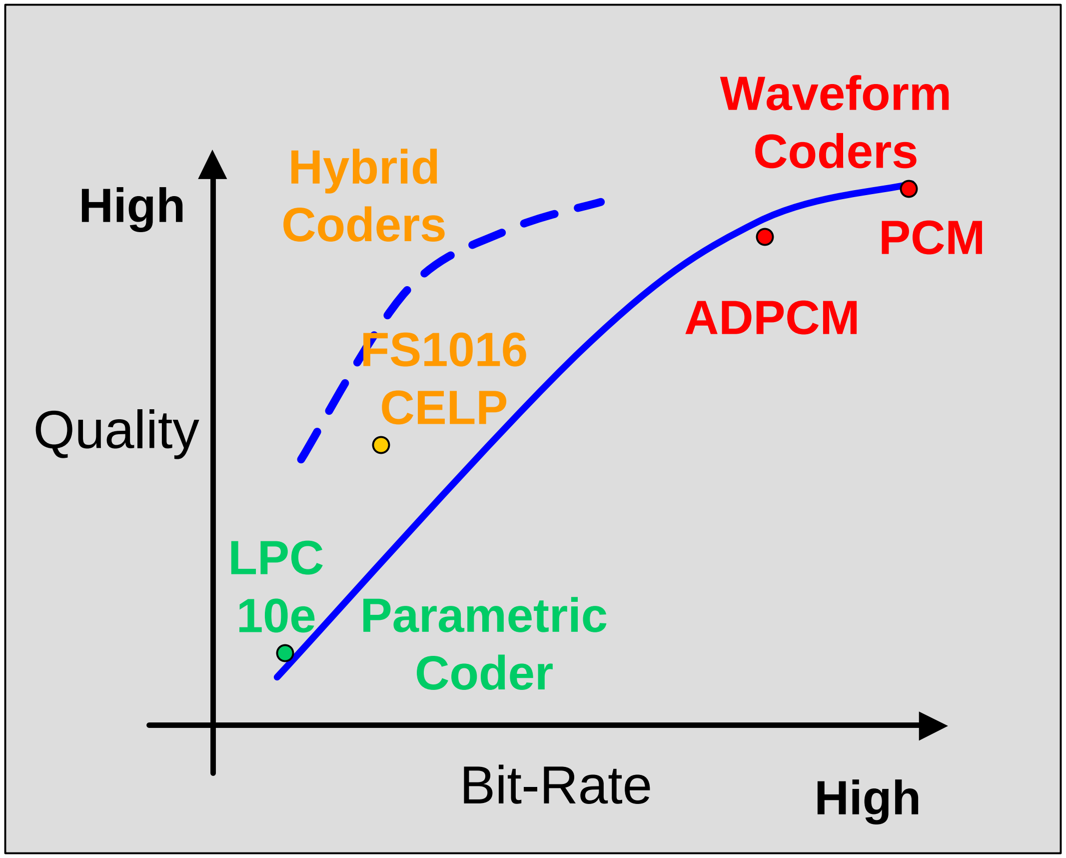


Speech Coding Criteria



Depending on the end-use, each criterion requires a different weightage.

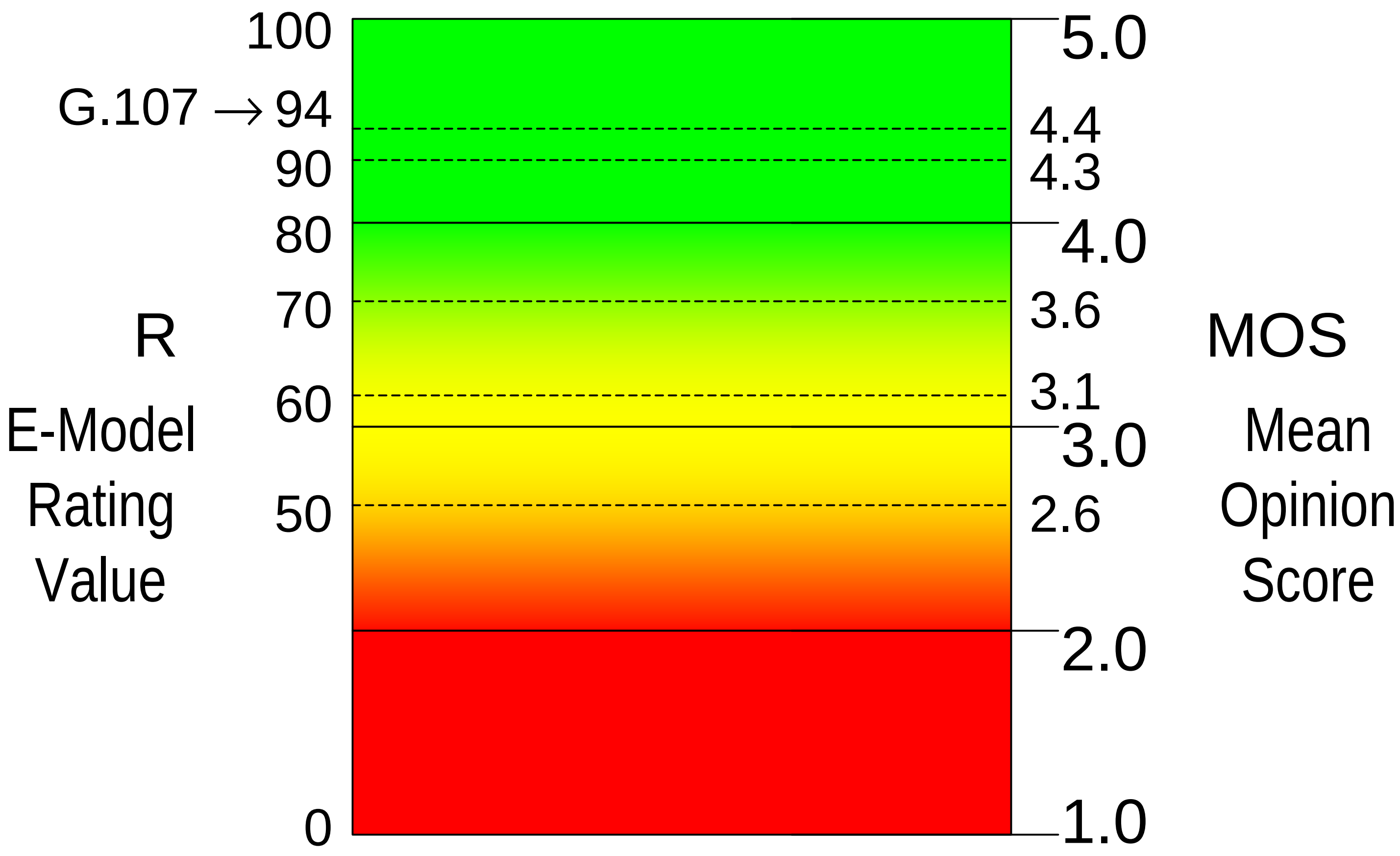
Speech Coding Approaches



Hybrid Codecs employ tools such as vector quantization to gain quality at low bit rates.

Hybrid codecs are more costly to implement.

Voice Quality



Some Voice Codec Options

Codec	Bit Rate (kbits/s)	Delay	Notes
G.711 (PCM)	64	0.125ms	Toll quality
G.726 (ADPCM)	32	0.25ms	Toll Quality
G.728	16	0.625ms	Licensing required
G.723.1	5.3 / 6.3	30ms (7.5ms)	Licensing required
G.729 / G.729A	8	10ms (5ms)	Licensing required
GSM (RPE-LTP)	13	20ms	Patents? Low Quality
iLBC	13.4 / 15.2	30ms / 20ms	Royalty-free / IETF
Speex	2.2 - 44	30ms	Open source

NOTE: Delay column shows frame size T_F and look-ahead buffer duration T_{LA}, if any.
Total codec processing delay is $2_{TF} + T_{LA}$

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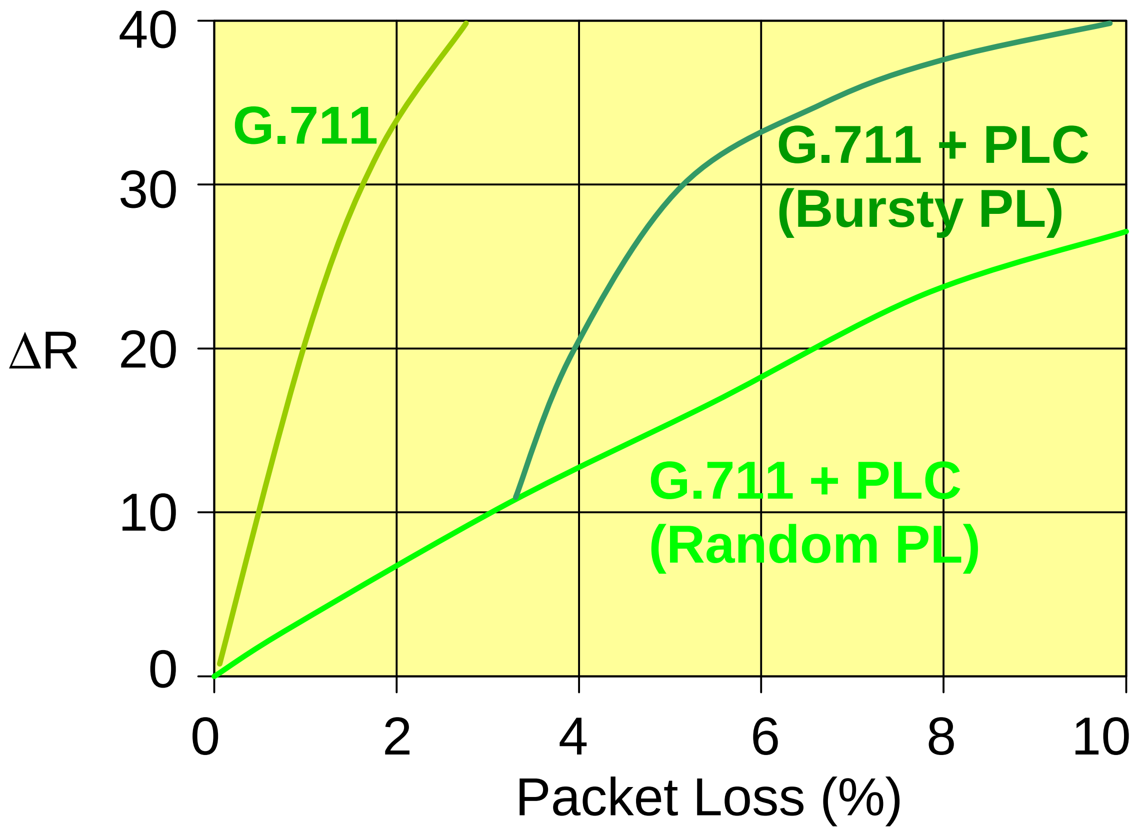
Voice Codec Costs

Codec	Bit Rate (kbits/s)	DSP Processing Budget
G.711 (PCM)	64	< 1 MHz (with PLC)
G.726 (ADPCM)	32	4 – 9 MHz (with PLC)
G.728	16	20 – 30 MHz
G.723.1	5.3 / 6.3	12 – 20 MHz
G.729 / G.729A	8	12 – 20 / 6 – 10 MHz
GSM (RPE-LTP)	13	3 – 8 MHz
iLBC	13.4 / 15.2	8 – 12 MHz
Speex	2.2 - 44	?

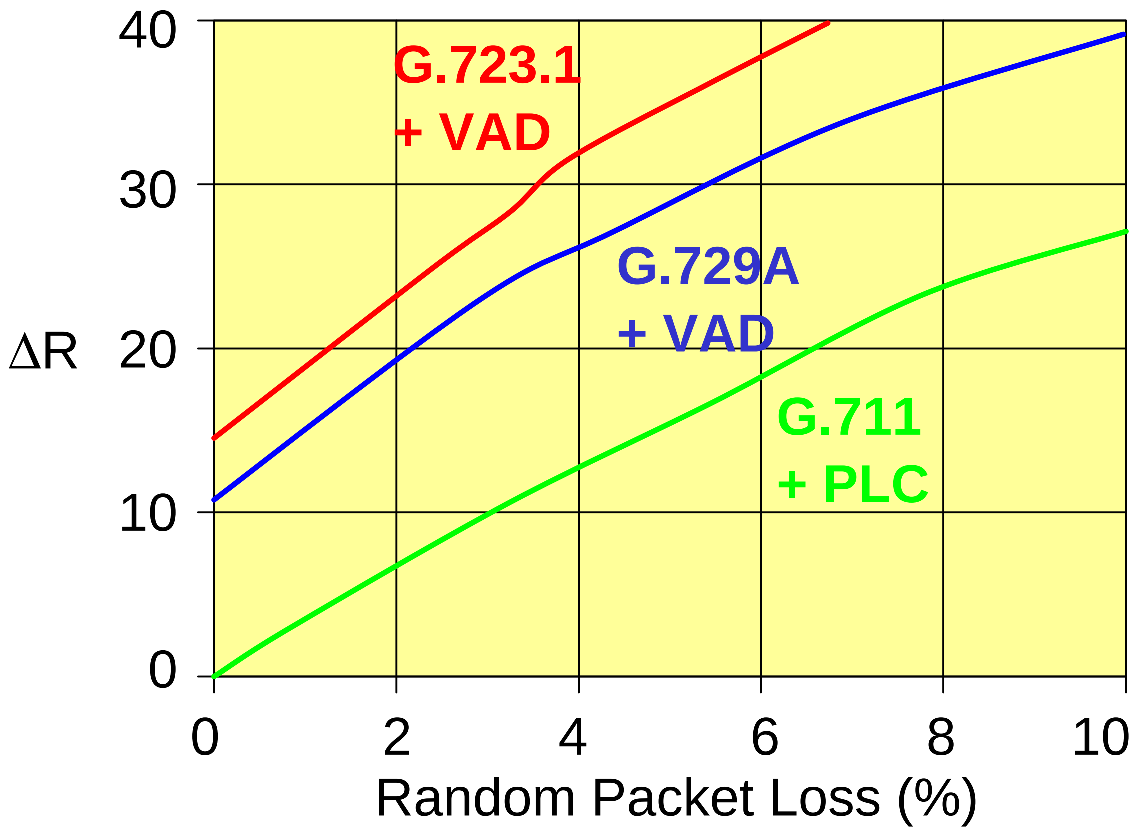
NOTE: Program / data memory requirements of each codec should also be considered.

R-degradation: Packet Loss



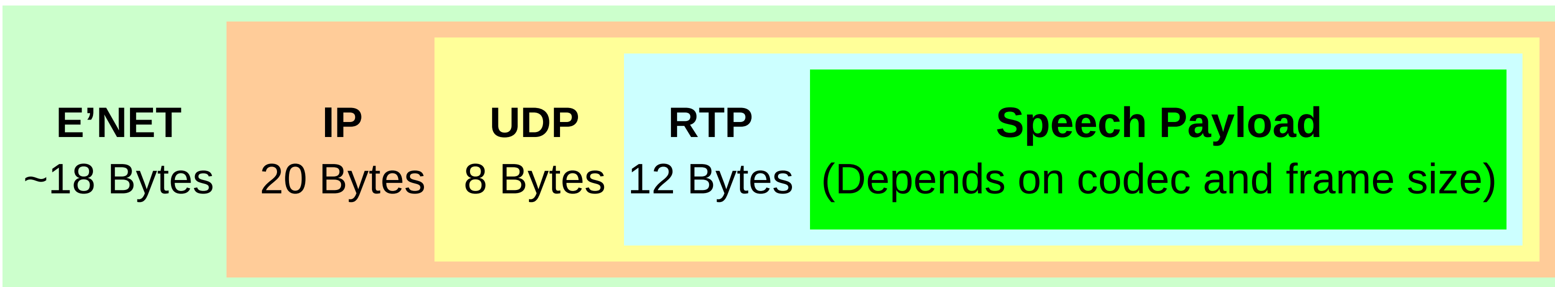
Source: Luis F Ortiz (Brooktrout Technology), RTC Magazine, July 2001

R-degradation: Packet Loss



Source: Luis F Ortiz (Brooktrout Technology), RTC Magazine, July 2001

Packet Header Overheads



- Packetization overheads can be significant.
- Header compression (cRTP) is possible for IP/UDP/RTP. Use of cRTP needs low round-trip delays (for header repair requests). This is primarily useful for (low-speed) local links.
- Large packets amortize overheads at the cost of extra latency.
- Other overheads (such as RTCP) are not counted here.

Bit Rates With Overheads

Voice Payload (Bytes) :

	Frame Size		
Codec:	10ms	20ms	30ms
G.711	80	160	240
G.726	40	80	120
G.729A	10	20	30

Channel Bit Rate (kbits/s) :

	Frame Size		
Codec:	10ms	20ms	30ms
G.711	110.4	87.2	79.5
G.726	78.4	55.2	47.5
G.729A	54.4	31.2	23.5

Unless packet header overheads are reduced, benefits of low bit-rate codecs are not fully utilized.

Signal Processing Options

- Low-channel CPEs
  - One General purpose µP alone
  - One DSP alone
  - Single-chip IP Phone SoC (µP + DSP + I/O)
  - One uP (host) + one or more DSP(s)
- Large CPEs, Gateways
  - Multiple hosts + DSP Farms
  - Hosts + DSPs with HW Accelerators
  - Few hosts + SoC (Processors + HW + I/O)

Signal Processing Costs

- DSP MHz numers below typically scale up by 1.5X to 3.0X on general purpose processors.
- Only functions that contribute to peak processor load are listed.
- Memory usage (not presented) is often a critical factor.

Functions	DSP MHz / voice channel
Codec (VAD-CNG, PLC)	**3 – 30 (depends on codec choice)**
G.168 EC	**3 – 10 (depends on EC design)**
DTMF Tx + Rx	**1 – 4**
Caller ID Tx	**1 – 2**
Other Functions	**4 – 14 (Jitter buffer, packet processing, I/O handling, task scheduling**
Total:	12 – 60 MHz

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Trends & Issues

- Wide band (7 kHz) voice codecs
- Stereo audio (conferencing) (?!?)
- Improved multi-party conferencing support
  - Conference bridges with multi-casting?
- Improved QoS
- Improved security
- Lower power consumption

In particular, the emerging VoWLAN (or VoWiFi) market needs this support.

Packet Voice References

- Books
  - F. Ohrtman, "Voice over 802.11", Artech (2004)
  - A. Sulkin, "PBX Systems for IP Telephony", McGraw-Hill (2002)
  - D. J. Wright, "Voice over Packet Networks", Wiley (2001)
- IETF
- Cable Labs (PacketCable)
- ATM Forum
- Frame Relay + MPLS Forum
- Historical papers (packet voice problem not new):
  - W. A. Montgomery: "Techniques for packet voice synchronization" IEEE JSACS, SAC-1, 6, Dec 1983
  - J. Gruber, L. Strawczynski, "Subjective effects of variable delay and speech loss in dynamically managed systems", IEEE Globecom (1982).