Voice over IP overview

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Oct. 2002

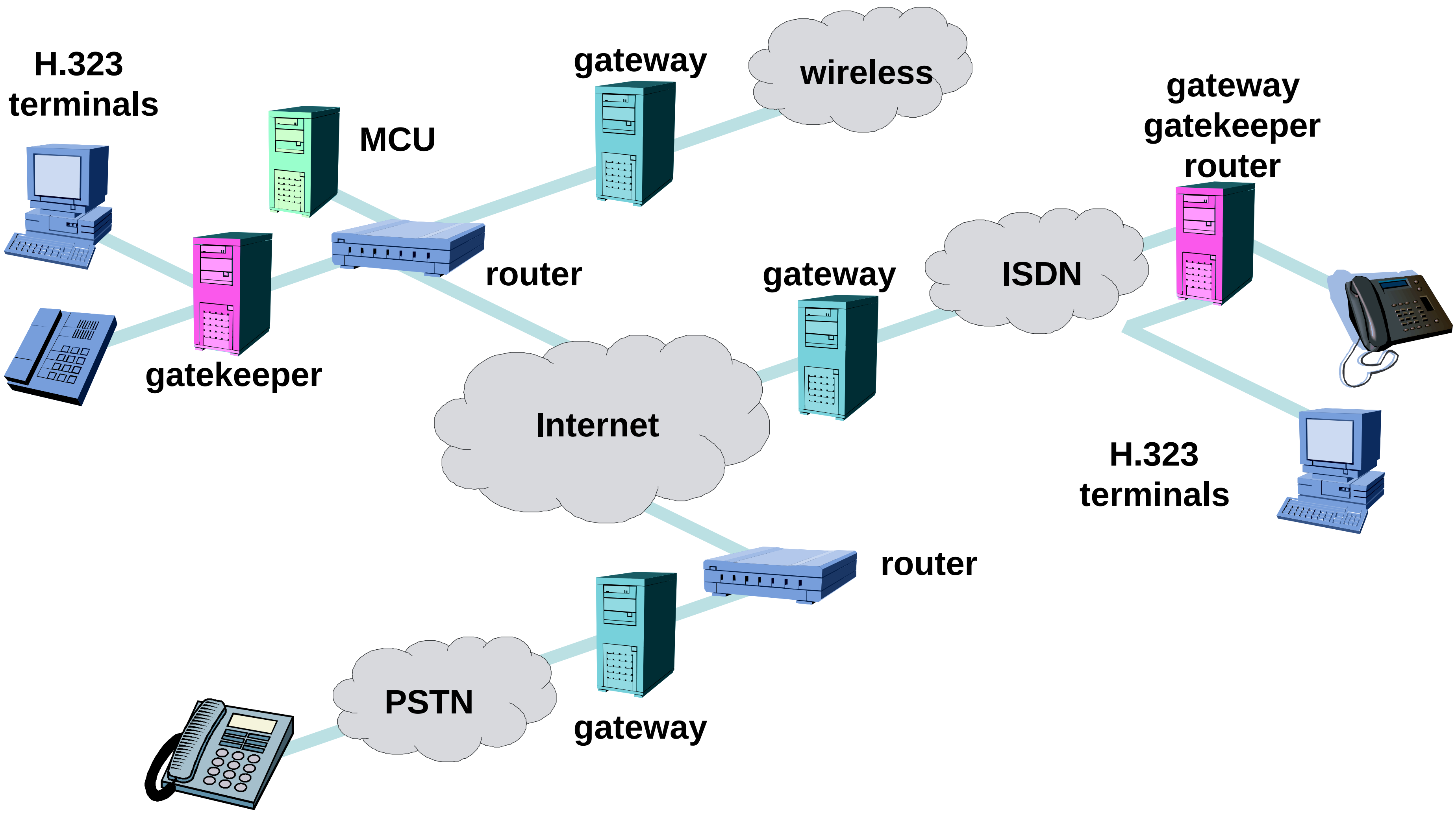
- **VoIP context**
- Int. Telecomm. Union (ITU) VoIP framework
- Key computation: audio codec
- Optimization goals and constraints
- IETF activities
- Available tools/libraries and products
- VoIP platforms
- Implications on a CSA for VoIP

Why use VoIP?

- Better bandwidth utilization by:
  - Using compression
  - Exploiting silence periods during conversations
  - Sharing of equipment for voice and data traffic (unified processing)

- Introduction of new services:
  - Conferences, distance learning, etc.

VoIP application scenarios



Elements of a VoIP network

- Terminals: LAN-based communication end-points
- Gateway (media and/or signaling):
  - Interface between packet- and circuit-switched networks
    - ? Media gateway: voice transcoding, protocol conversion
    - ? Media gateway controller: call handling, call state
    - ? Signaling gateway: signaling mediation
- Gatekeeper:
  - Admission control, SNMP services, address translation
- MCU: Multipoint Control Unit:
  - Handling of broadcasts / conference calls

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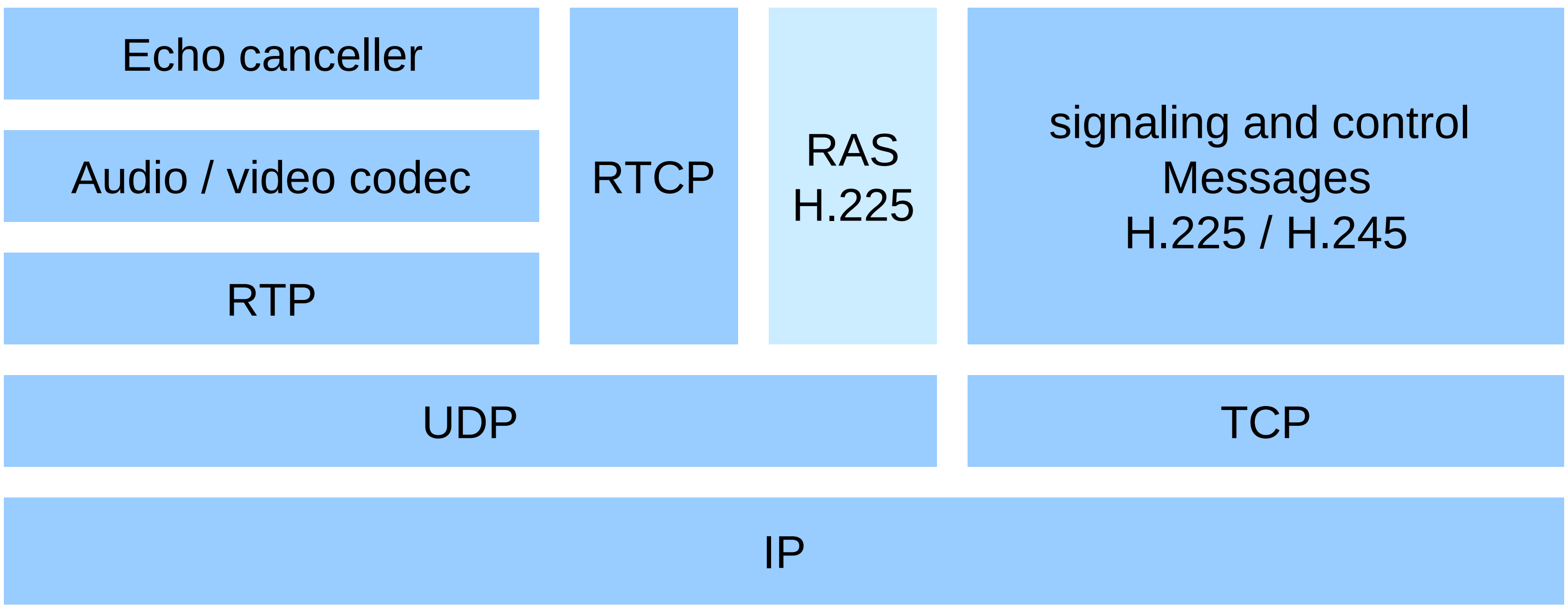
ITU-T H.323

- Audio Codecs: G.711, G.722, G.723, etc.
- Call control, signaling: H.225, Q931
- Path and parameter negotiation: H.245

- Additions:
  - Video Codecs: H.261, H.263, etc.
  - Security, encryption, authentication: H.235
  - Supplementary services: H.450 (call transfer, call waiting, etc.)

- Gatekeeper and media gateway:
  - RAS: registration, admission, status: H.225
  - Megaco: H.248 media gateway control protocol

H.323 protocol stack



H.323 call phases

- Phases 1 and 7 only present if gatekeeper (GK)-routed call.

Phase		Protocol	Intended functions
1	Call admission	RAS	Request permission from GK to make/receive a call. At the end of this phase, the calling endpoint receives the Q.931 transport address of the called endpoint.
2	Call setup	Q.931	Set up a call between the two endpoints. At the end of this phase, the calling endpoint receives the H.245 transport address of the called endpoint.
3	Endpoint capability negotiation and logical channel setup	H.245	Negotiate capabilities between two endpoints. Determine master-slave relationship. Open logical channels between two endpoints. At the end of this phase, both endpoints know the RTP/RTCP addresses of each other.
4	Stable call	RTP	Two parties in conversation.
5	Channel closing	H.245	Close down the logical channels.
6	Call teardown	Q.931	Tear down the call.
7	Call disengage	RAS	Release the resources used for this call.

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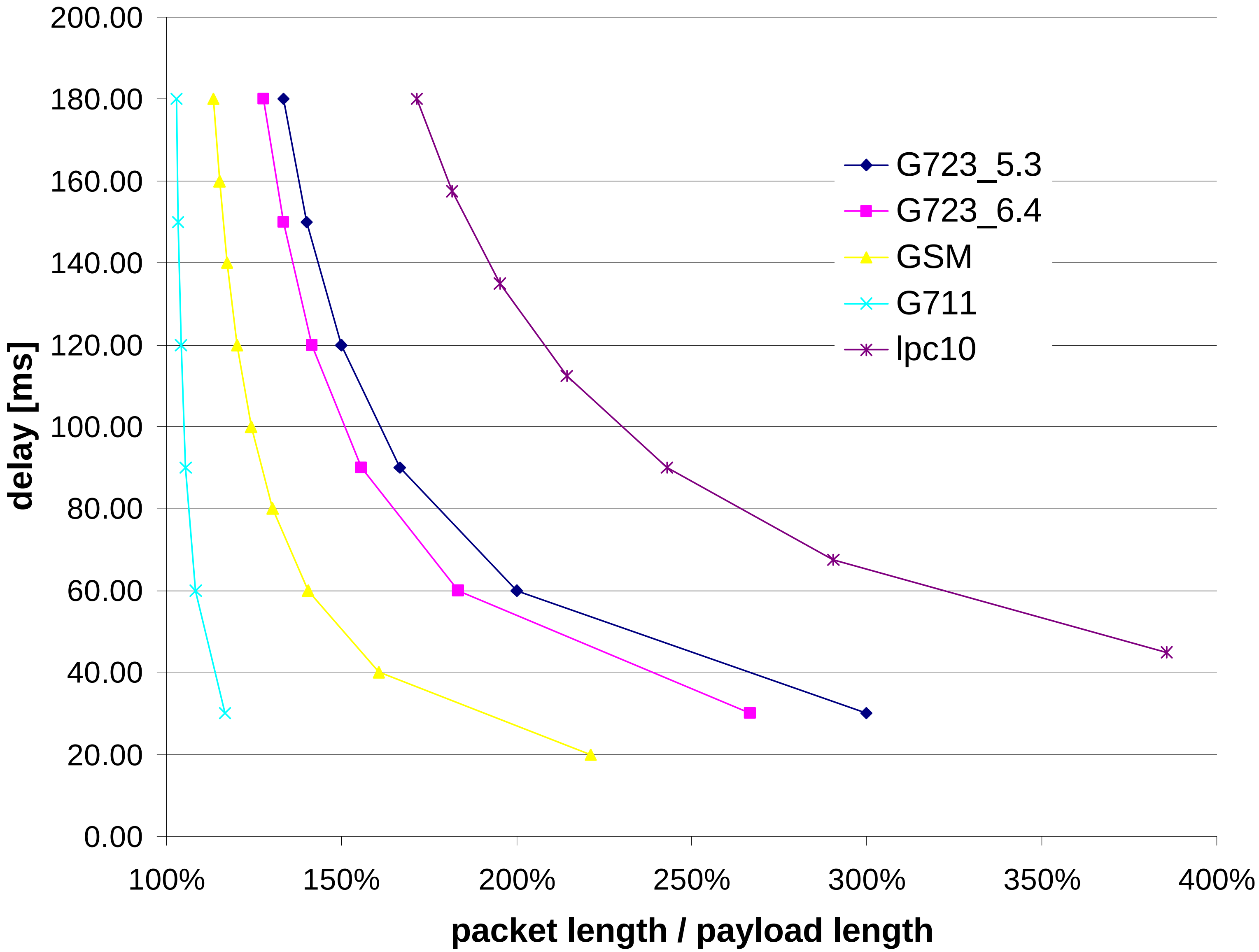
Audio codec choices

ITU no.	Rate (CBR) [Kbit/s]	algorithm	Frame length
G.711	64	Pulse Code Modulation (PCM), 8000 samples/s, 8 bit resolution	0.125 ms
G.723.1	6.4 / 5.3	multipulse maximum likelihood quantization / algebraic-code- excited linear prediction	30 ms
G.726	40 / 32 / 24 / 16	adaptive differential PCM	0.125 ms
G.728	16	low-delay code-excited linear prediction	2.5 ms
G.729	8	conjugate-structure algebraic-code-excited linear prediction	10 ms
GSM	13.2	regular-pulse excitation long-term prediction	20 ms
LPC10	2.5	Linear predictive coding	22.5 ms

Packet delay vs. network utilization

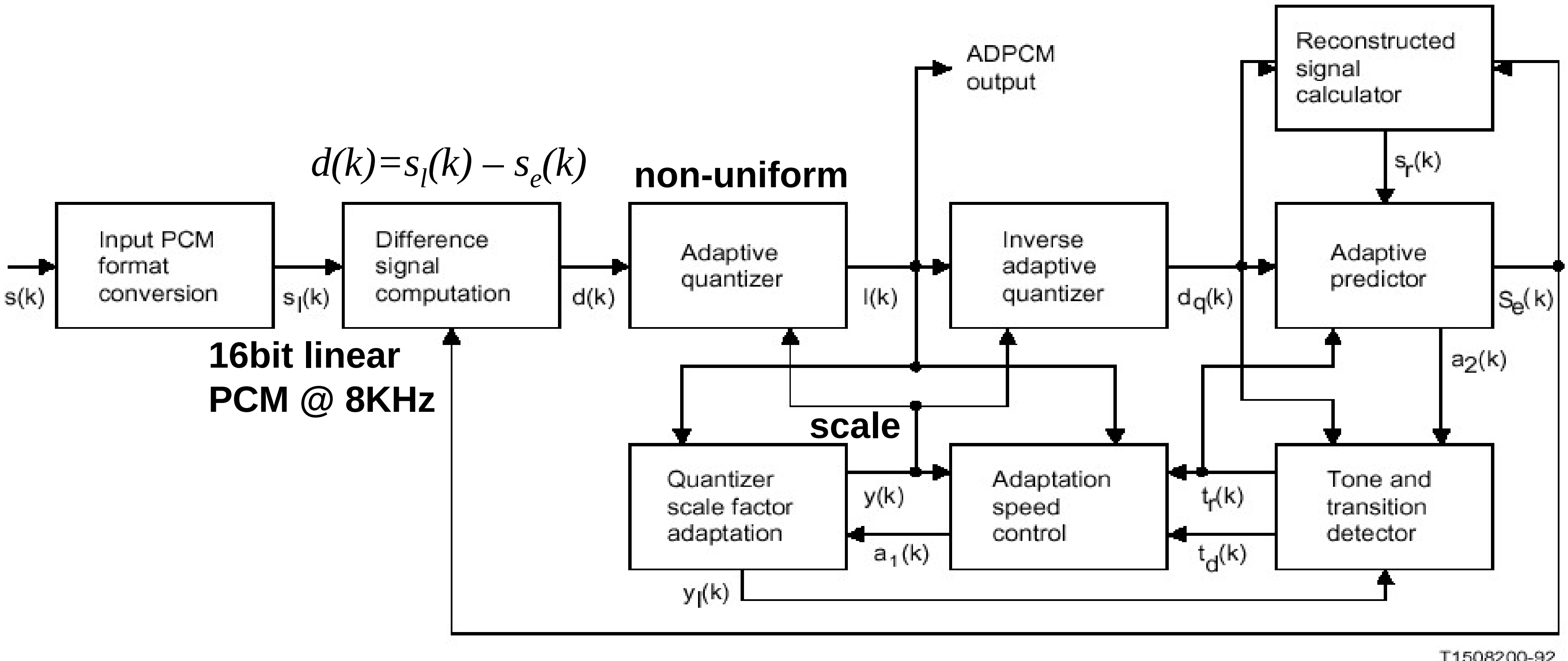


Complexity vs. code size vs. quality

- On Hitachi SH3-DSP [Yashvant Jani, www.embedded.com]
- Voice quality Mean Opinion Score (MOS):
  (1: bad, 4: toll quality, 5: excellent)

codec	MIPS	code size	MOS
G.711	<1	<1KB	4.4
G.721	10.0	2.1KB	
G.722	10.0	2.6KB	
G.723.1	29.7	56KB	3.6
G.729	29.5	35KB	3.9
G.729a	16.5	32KB	4.0

G.726 encoder block diagram



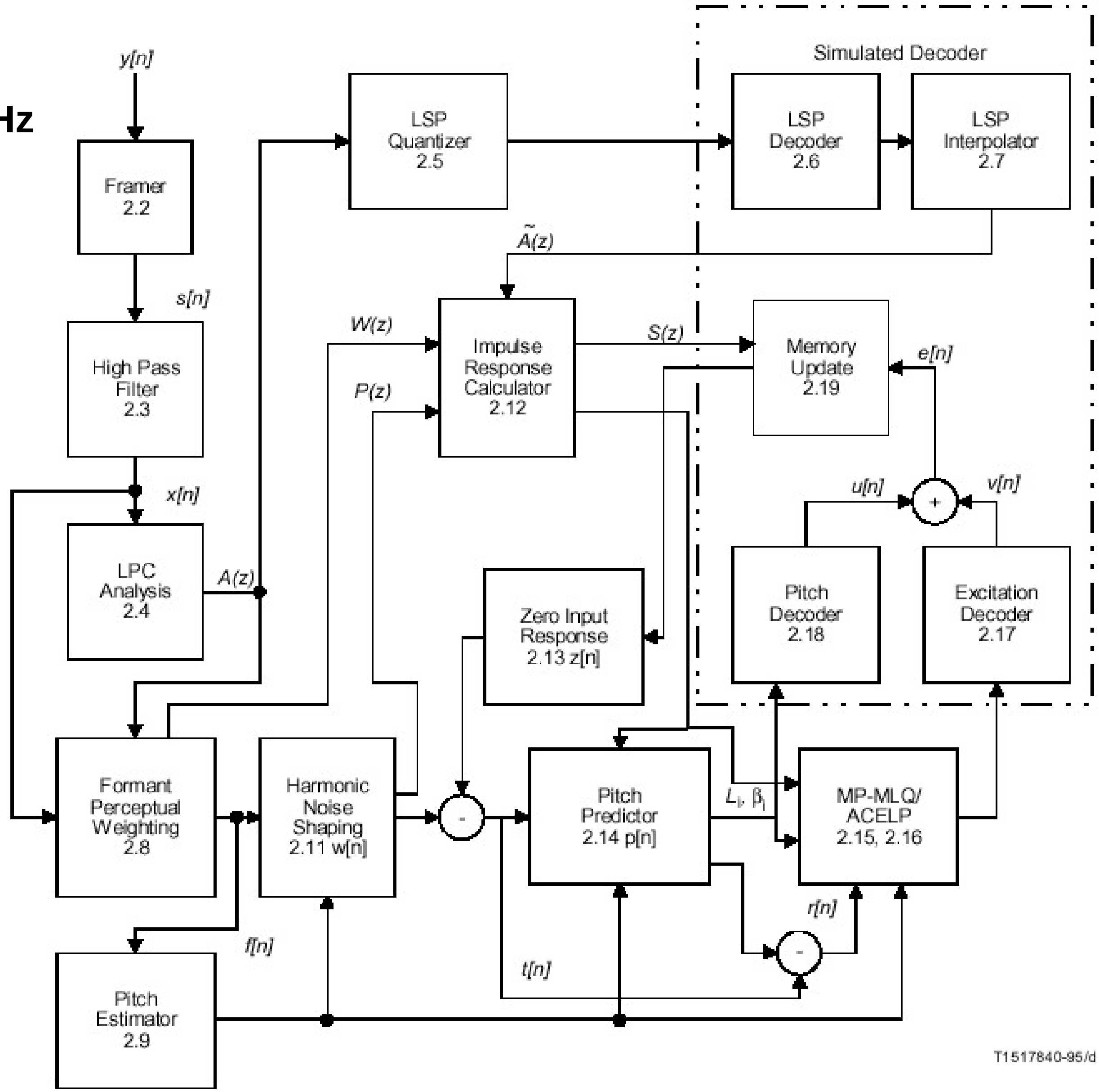
$$y_u(k) = (1-2^{-5})y(k)+2^{-5}W[I(k)]$$

$$y_l(k) = (1-2^{-6})y_l(k-1)+2^{-6}y_u(k)$$

$$y(k) = a_l(k)y_u(k-1)+(1-a_l(k))y_l(k-1)$$

G.723.1 encoder block diagram

16bit linear
PCM @ 8KHz



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Constraints

- One way end-to-end latency budget: 150ms
  (ITU-T G.114)

- Signal quality:
  - Echo cancellation performance tests
    (ITU-T G.168, G.165)
  - Toll quality Mean Opinion Score (MOS, ITU-T P.800)

- Voice packet loss < 5%; less if grouped in bursts

- Call setup time < 30s

- SS7 over IP ITU constraints:
  - Response time < 1.2s
  - Availability of signaling route > 99.9998%
  - SS7 message loss: 1 in 10^7 max.
  - Out of sequence delivery: 1 in 10^{10} max.
  - Erroneous messages: 1 in 10^{10} max.

Components of end-to-end latency

- Coder frame length
- Coding / Decoding processing time
- Network transmission: queuing delay, transport delay, variable delay due to network conditions
- Jitter compensation (typ. 80 ms)

Gateway optimization goals

- Power per channel
- Cost per channel
- Channel density (e.g. per in^2)
- Functionality
- Performance:
  - max. call setup rate, processing latency, availability, call success ratio
- Equipment classes:
  - T1 (24 channels), E1 (30), DS3 (672), OC3 (2016)

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Competing IETF drafts:

- Session Initiation Protocol (SIP, rfc2543)
  - Name translation, user location (SIP URLs)
  - Feature negotiation
  - Call participant management during session
- Media Gateway Control Protocol (MGCP, rfc2705)
- Stream Control Trans. Protocol (SCTP, rfc2960):
  - SS7 over IP

IETF working groups, SIP related

- PINT / SPIRITS:
  - Interacting telephony services (PSTN and Internet)
- ENUM:
  - Number resolution using DNS
- TRIP:
  - Advertising reachability and route attributes between administrative domains (e.g. service providers)
- SIGTRAN:
  - Signal transport working group

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Available audio source code

- ITU:
  - Reference implementations for audio codecs (int, fixed-p., and floating-p.): G.722, G.723.1, G729
- TU-Berlin:
  - GSM codec
- LPC10 from several sources

Available source code: ITU H.323

- openh323.org, gnugk.org:
  - gatekeeper, gateway, and terminal software (including G.711 audio)
- Gnomemeeting.org:
  - H.323 gatekeeper and terminal software (including LPC10, GSM, G.711, G.726 audio)

Available source code: IETF SIP

- linphone.org:
  - SIP client (including LPC10, GSM, G.711)
- asterisk.org:
  - SIP and H.323 terminal and gateway (including GSM, G.711, G.726 audio)

Libraries:

- IBM Alphaworks j323:
  - H.323 terminal functions in java
- vovida.org:
  - SIP server functions (provisioning, policing, redirection, H.323 protocol translation, etc.)
- RadVision.com, trillium.com:
  - H.323 terminal & gatekeeper, SIP terminal & server APIs
- openh323.org, iptel.org, h323forum.org lists:
  - 22 commercial and non-commercial H.323/SIP protocol stacks

Products: listed @ openh323.org, iptel.org, h323forum.org

- 31 software H.323 clients, “softphones”
  (MS, Intel, Cisco, Sun, Netscape, etc.)
- 21 SIP softphones (Nortel, Microsoft, Siemens, etc.)

- 26 H.323 hardphones (Cisco, Siemens, etc.)
- 15 SIP hardphones (3Com, Siemens, Nortel, Cisco, etc.)

- 22 SIP-enabling server software packages
- 32 SIP gateways, 57 H.323 gateways

- 21 H.323 gatekeeper

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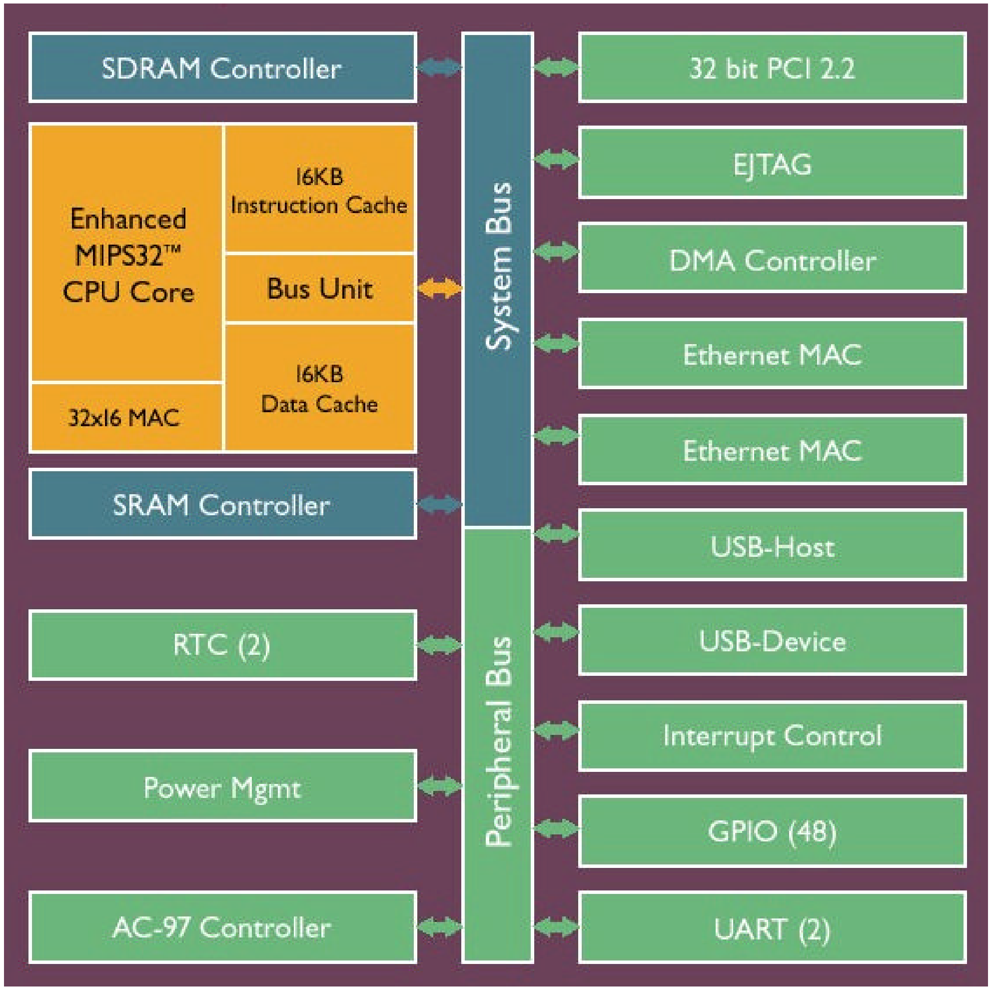
IP telephone/ small office gateway

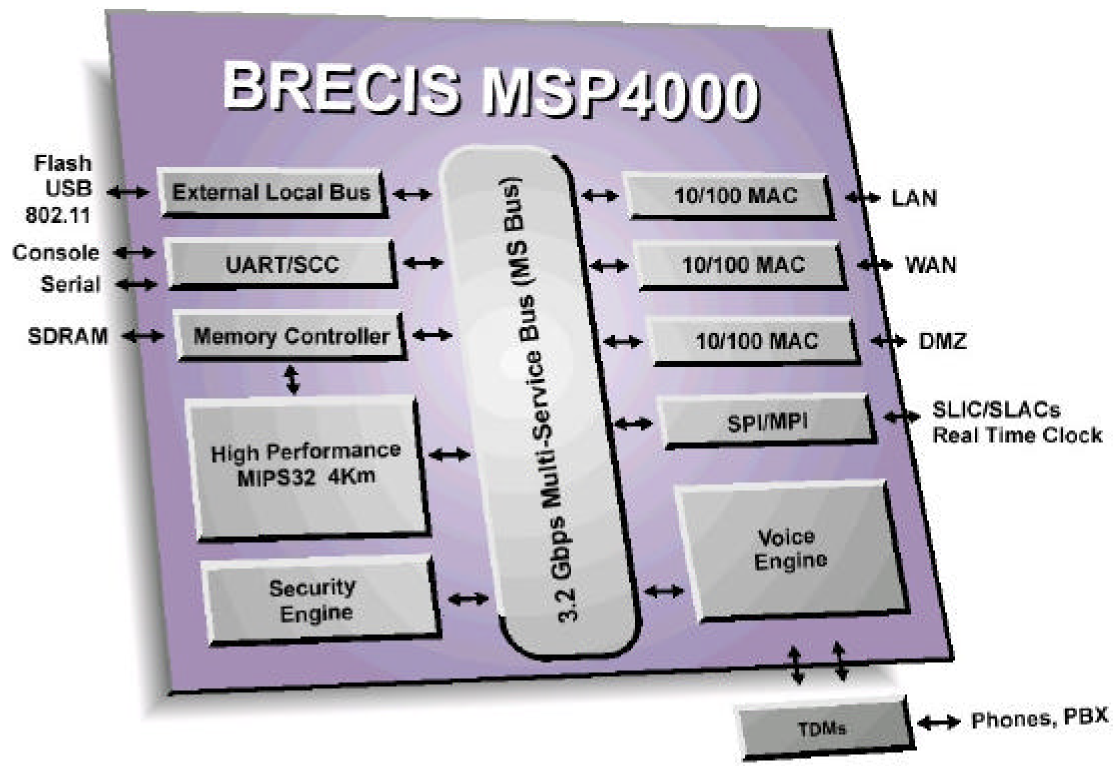
- One to 16 PCM channels
  - AMD Alchemy Au1500
  - Brecis MSP4000, MSP5000
  - Agere Phone-On-A-Chip
  - Netergy Audacity T2
  - Broadcom BCM1101

Chips: AMD Alchemy Au1500

- MIPS32 core,
  16KI, 16KD, @500 MHz
- Two fast Ethernet ports; PCMCIA, PCI, USB, SRAM, SDRAM controllers
- MAC unit, MMU
- Windows CE, VxWorks, Linux support

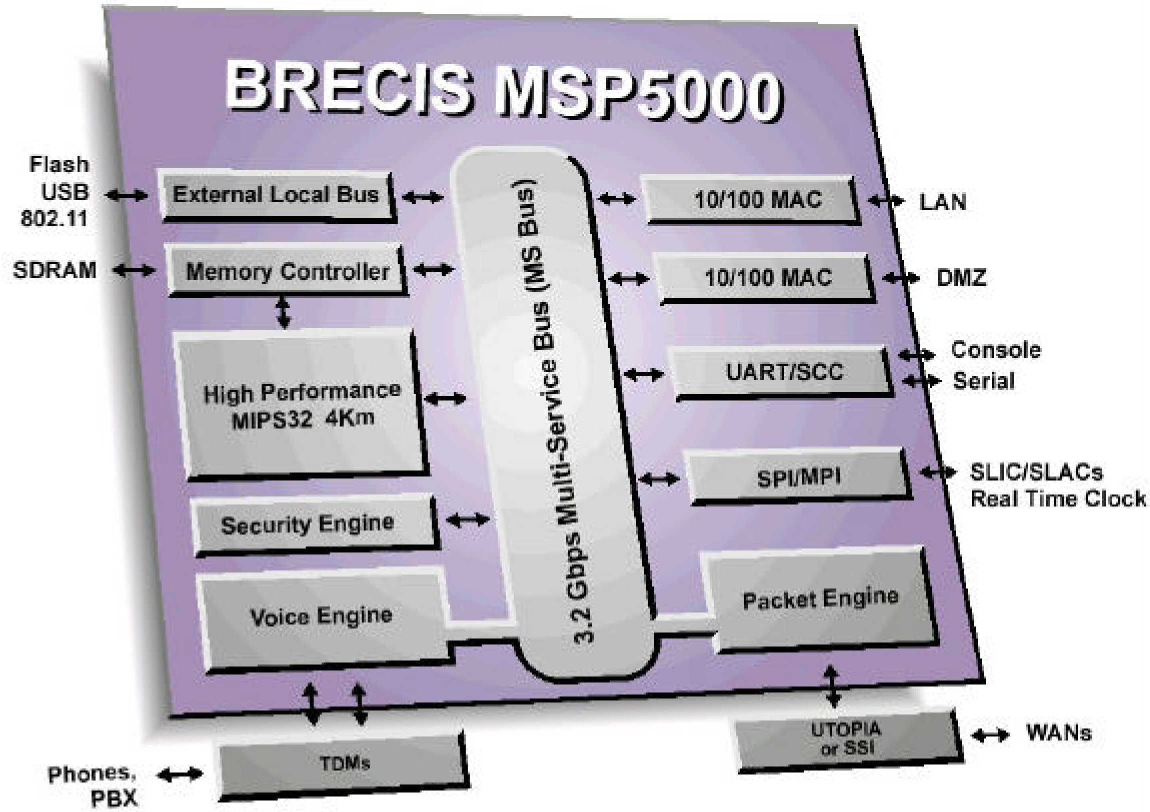


Brecis MSP4000



- MIPS32-based, 16KI, 16KD, @150MHz
- Security engine: 40MBit/s 3DES
- SDRAM controller
- Four voice channels
- Three fast Ethernet ports
- Voice engine: dual-MAC LSI ZSP DSP core @100MHz, 80KI, 64KD, ADPCM in HW
- VxWorks, Linux support; APIs

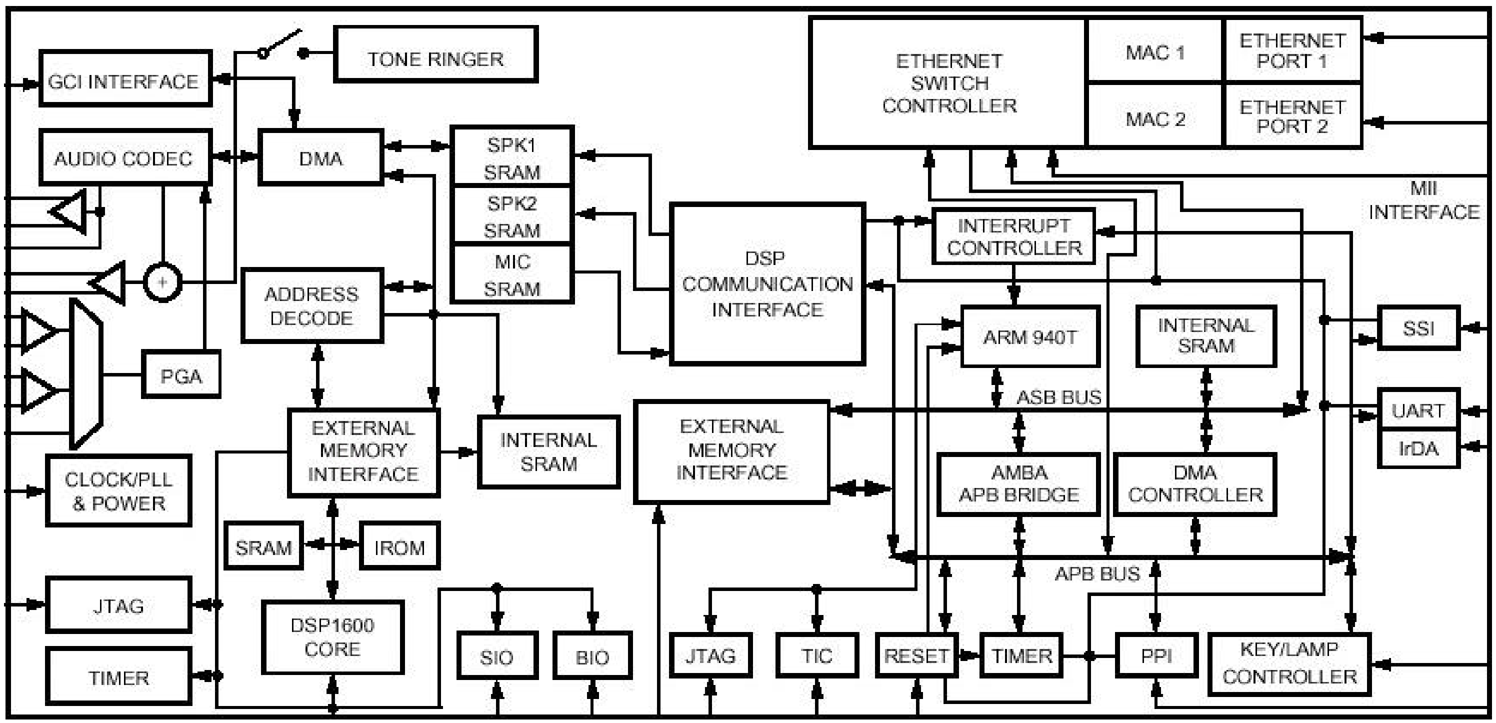
Brecis MSP5000



- MIPS32-based, 16KI, 16KD, @150MHz
- SDRAM controller
- 16 voice channels
- Two fast Ethernet ports
- Security engine
- voice engine: dual-MAC LSI ZSP DSP core @125MHz, 80KI, 80KD, ADPCM in HW
- Packet engine: 2nd LSI ZSP DSP (w/o ADPCM)
- VxWorks, Linux support; APIs

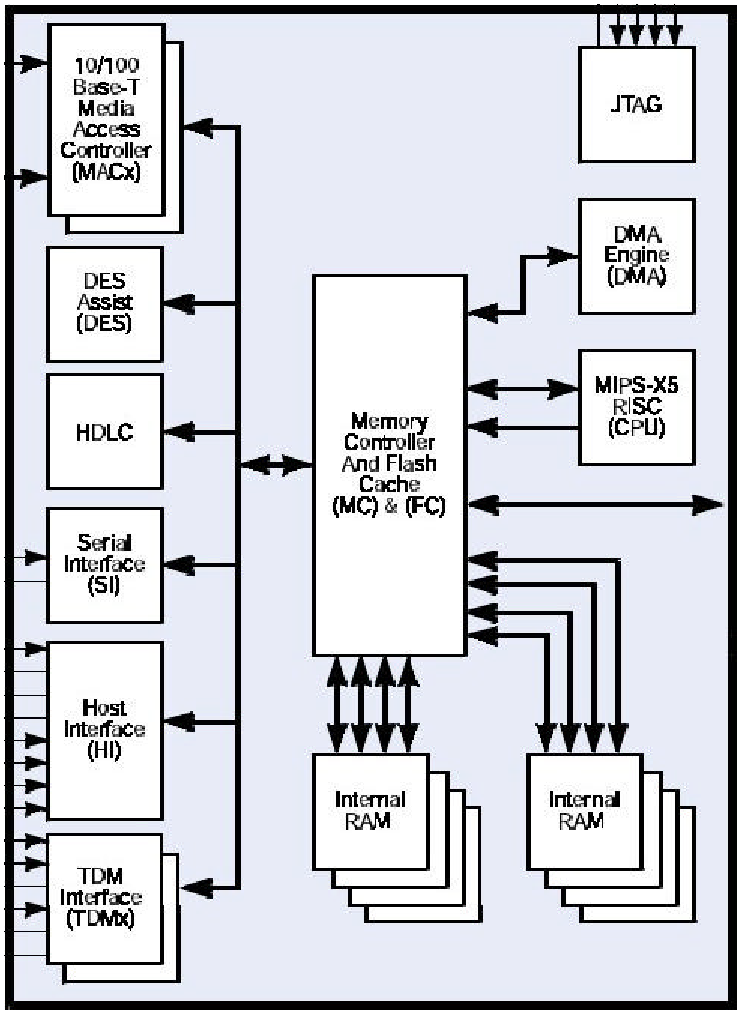
Agere Phone-On-A-Chip

- Agere DSP 1600 @100MHz
- ARM940T core @70MHz, 4KI, 4KD
- Two ethernet ports
- A/D, D/A converters
- Key and lamp controller
- VxWorks support, Trillium H.323 stack

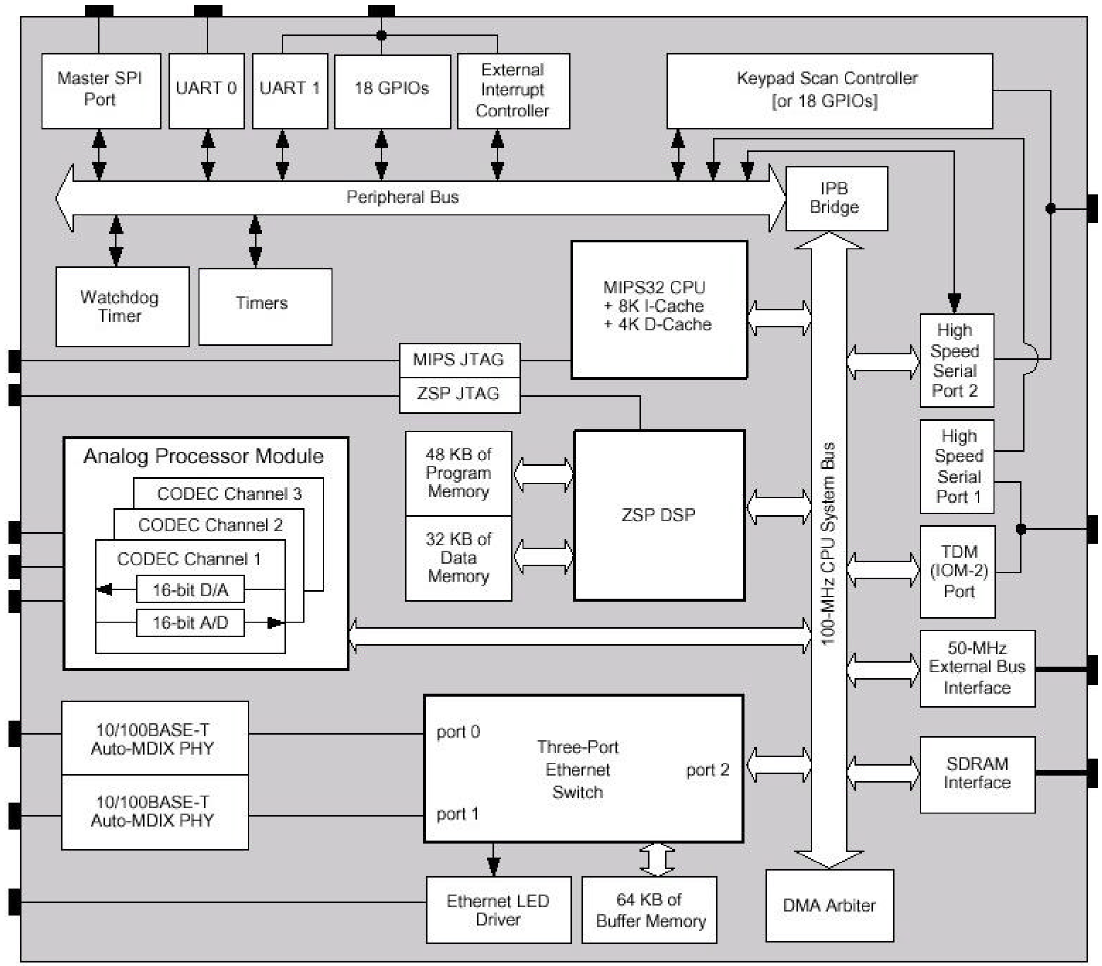


Netergy Audacity T2/T2U

- MIPS core with MAC
- 256 KByte SRAM
- Two fast Ethernet ports
- DES unit
- T2U: four voice channels, 384KByte SRAM
- POSIX OS with proprietary H.323/SIP stack



Broadcom BCM1101



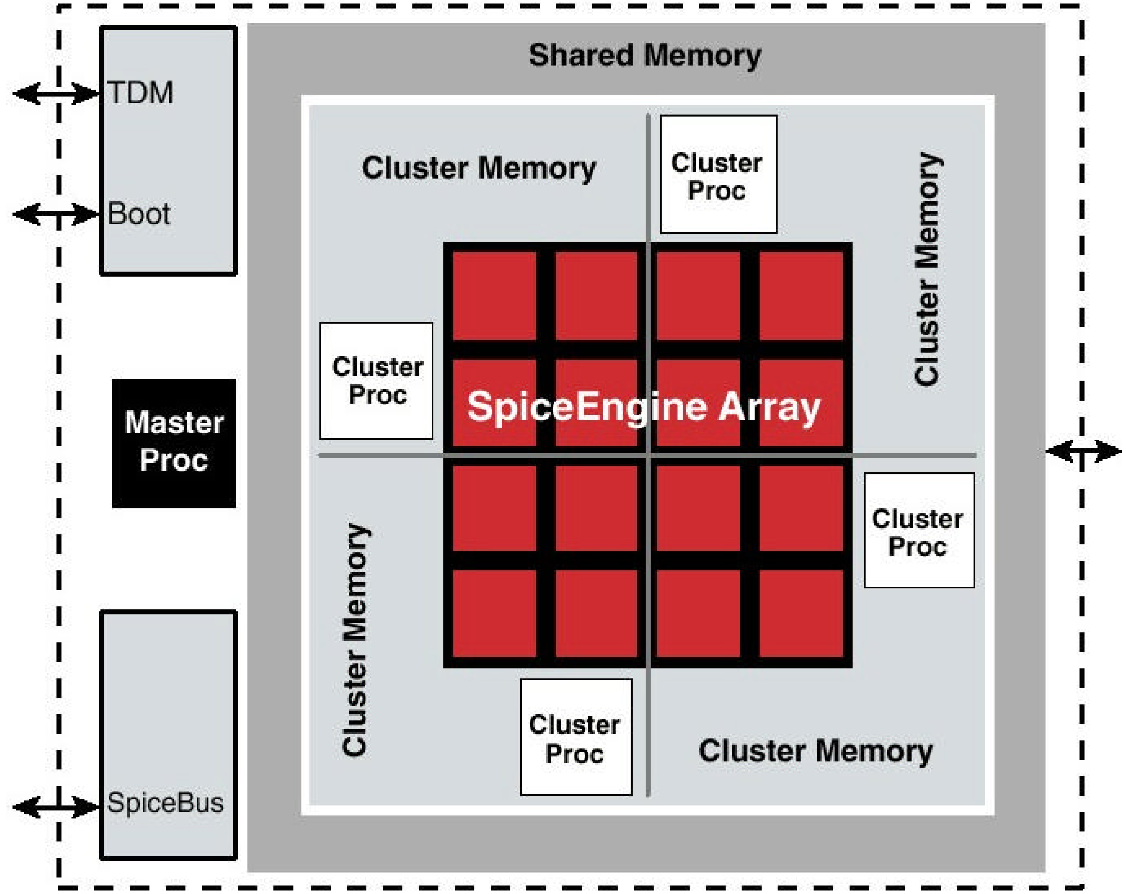
- MIPS32, 8KI, 4KD, @150MHz
- Two voice channels, three fast Ethernet ports
- Signal processing API
- dual-MAC LSI ZSP DSP core @140MHz, 48KI, 32KD

Gateway platforms

- Broadcom Calisto BCM 1500
- AudioCodes AC486

- Motorola C-Port plus several Motorola MSC8102 DSPs
- Intel IXP plus several Intel IXS1000 media processors

Broadcom Calisto BCM 1500



- 240 PCM channels
- 768KByte shared mem
- 16 DSPs (four clusters)
- 5 Xtensa CPUs @ 166MHz, MAC, 4KI, 4KD

AudioCodes AC486

- H.323 gateway
- 48 PCM channels

DSP examples:
(with VoIP SW support)

- Motorola MSC8102:
  - Four StarCore DSPs @300MHz, 16KI, 224KD, 476K shared, 600 PCM channels, using C-Port for VoIP
- TI TNETV3010:
  - Six C55 DSPs @300MHz, 216 PCM channels
- ADI ADSP2192:
  - Dual core @160MHz, 12 channel VoIP
- BOPS VoiceRay core:
  - 36 PCM channels @200MHz
- Intel IXS1000: 240 channels, using IXP for VoIP
- 3DSP SP-5flex DSP core

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Implications on a CSA for VoIP

- Data (codec): Pipe and filter
  - Non-transforming connectors
  - Feedback loops
  - Asynchronous transforms of sets of voice samples

Implications on a CSA for VoIP

- Protocols: Layered system
  - Control topology: stack, partly lockstep, parallel
  - *Used for signaling:* sporadic, low volume, mode: passed
  - *Used for data:* continuous, high volume, mode: passed
  - Run-time identification of partner for transfer-of-control/data
  - Control/data direction: could be same as well as opposite

- Terminal – gatekeeper relation: client-server
- Terminal – terminal relation: communication processes:
  - Asynchronous: signaling
  - Synchronous: during call (regular exchange of state updates)

Appendix

Introductory references

- **Internet telephony: going like crazy,** *Thomsen, G.; Jani, Y.,* IEEE Spectrum , Vol.: 37 Issue: 5, May 2000

- **Voice over IP signaling: H.323 and beyond,** *Hong Liu; Mouchtaris, P.,* IEEE Communications Magazine, Vol.: 38 Issue: 10 , Oct. 2000

- **Voice over IP,** *Mehta, P.; Udani, S.,* IEEE Potentials, Vol.: 20 Issue: 4, Oct.-Nov. 2001