Hardware and Software Architecture of a Packet Telephony Appliance

Cormac J. Sreenan

AT&T Labs Research
Florham Park, NJ
cjs@research.att.com
http://www.research.at.com/~cjs

Collaboration with AT&T Labs colleagues:

Mike Chan, Chuck Cranor, R. Gopalakrishnan, Peter Onufryk, Larry Ruedisueli, Eric Wagner

http://www.research.att.com/~cjs/tops-project

Talk Outline

Packet telephony

- Background and previous work

Telephone appliance

- Design issues and principles

Hardware architecture

- Networked processor
- Telephone design

Software architecture

- Communications mechanisms
- Telephone application components

Conclusion

- Status, future work, related work

Packet Telephony Background

Using packet networks for voice communication

- Driven by potential for cost savings and new services
- Today mainly based on PSTN dial-in to gateway
- Gateways interconnected over IP networks

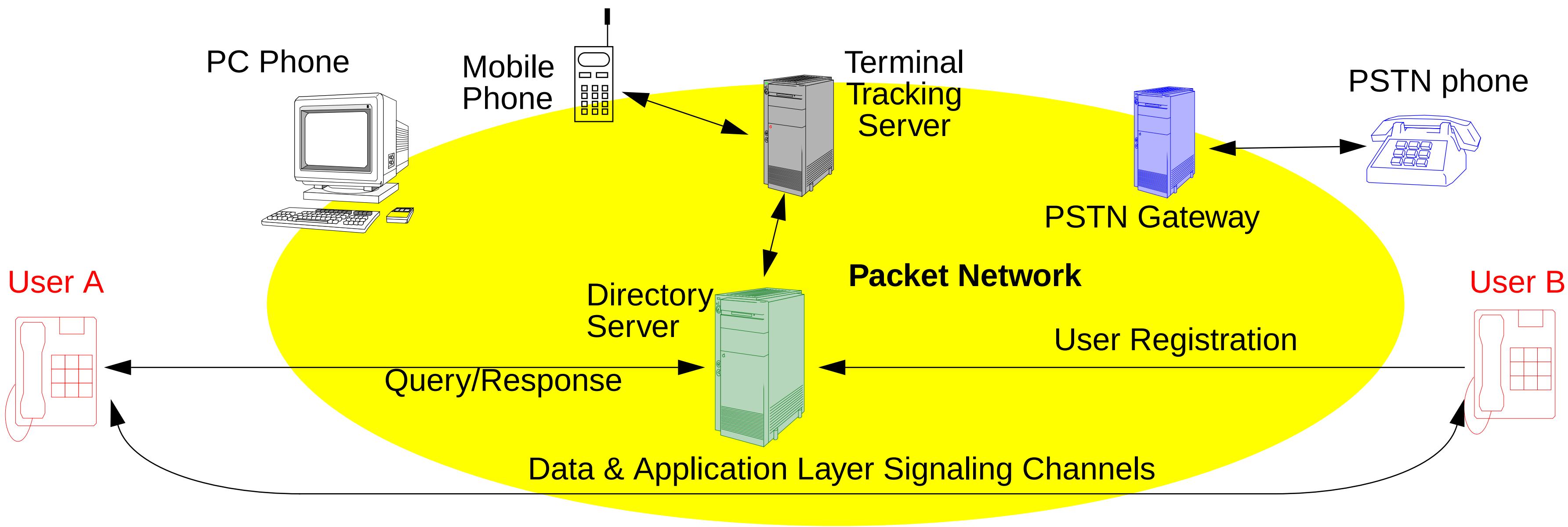
Need to examine consumer endpoint appliances

- Use of a office LANs for voice and data
- Penetration of home access and in-home networks

Intelligence at both the end-points and in the network

- Opportunity to re-partition functionality
- Application specific signaling supports telephony features (conferencing, teleporting, etc.)
- Flexibility to support new services and capabilities (e.g. multiple media)
- Implications for endpoint design

Previous work: TOPS Architecture (NOSSDAV'98)



Call Flow:

- User/Terminal Registration with Directory Server
- Directory Query (returns a call handling profile)
- Application Layer Signaling (ALS) establishes calls

Other servers: PSTN gateway, Terminal Tracking Server

Why Not a PC based Packet Phone?

Too Expensive ($500 too much for a phone)

Too Complex

- To use
- To install and configure
- To administer

Too Unsightly

- Big (keyboard + mouse + monitor + case)
- Loud (power supply fan, CPU fan, and disk)

Too Unreliable

- Never on when you need it
- Crashes often

Design Principles

Low cost (less than $100)

- A consumer device (a phone to put in your bedroom)

Extensibility

- Packet telephony is in its infancy -> standards changing
- Research areas still exist: QoS, security, privacy, billing
- Must be able to support new/advanced services

Ease of Use

- Designed for people with no technical background
- Ordinary people unwilling to invest time to set-up, configure, and maintain complex devices

Reliability

- Always on and always works

Euphony ATM Telephone (EAT)

Features:

- ATM-25 Network Connection
- RS232 (PC style DB9)
- Traditional telephone interface
- Case speaker and microphone
- External audio
  - Line output
  - Line / microphone input
- Two extra push buttons
- Three green status LEDs
- One red status LED

Why Not Include More "Stuff" with the Phone?

Things we could have added

- LCD display (touch screen), more keys, keyboard, mouse, ...

Adding "stuff" is bad

- Makes phone expensive and complex
- Would anyone really use it?
- How many ISDN phone features do you use?

Our approach for advanced features is a soft interface

- Phone runs web server
- Advanced features available via remote browser and other network devices
- You pay only for those features you want

Euphony Networked Processor

Like an Intel 486 DX2 66 with: signal processing instructions, audio interface, network interface, and system logic all for about $7

RISC Processor

- R4000 like (MIPS II) RISC processor

DSP Instructions

- Pipelined multiplier, Extract/Saturate instructions

ATM Interface

- single ATM Forum UTOPIA, single or dual AT&T DPI

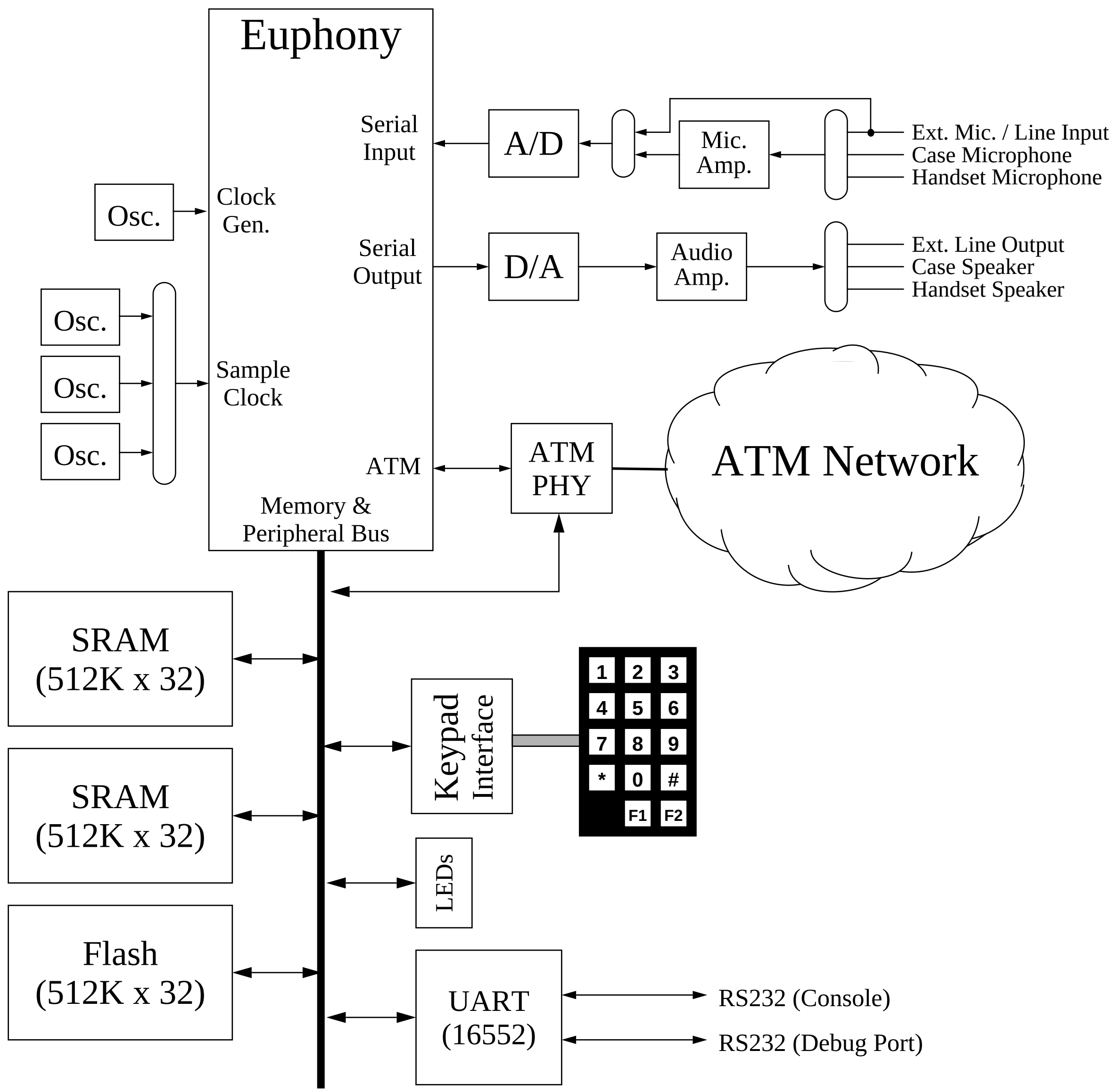
Digital Audio Interface

- Serial interface for D/A and A/D

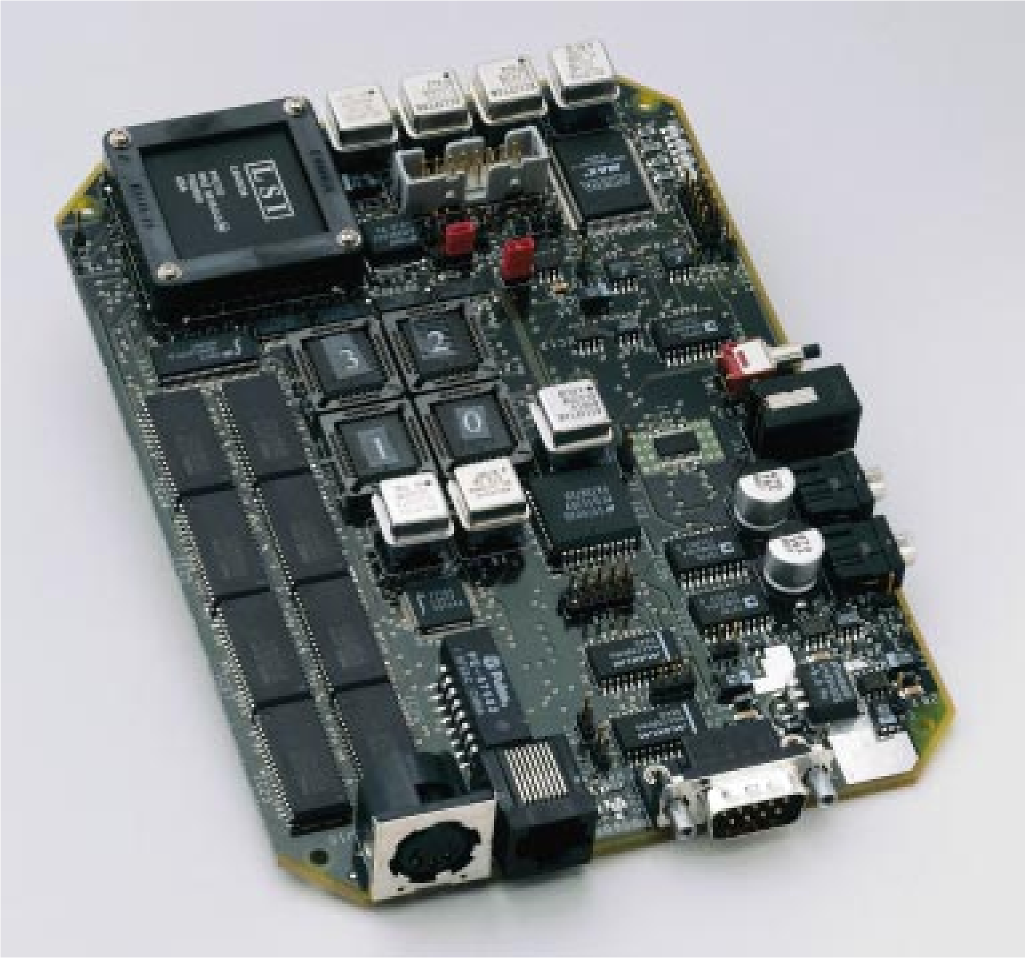
Support Logic

Low Power (500 mW)

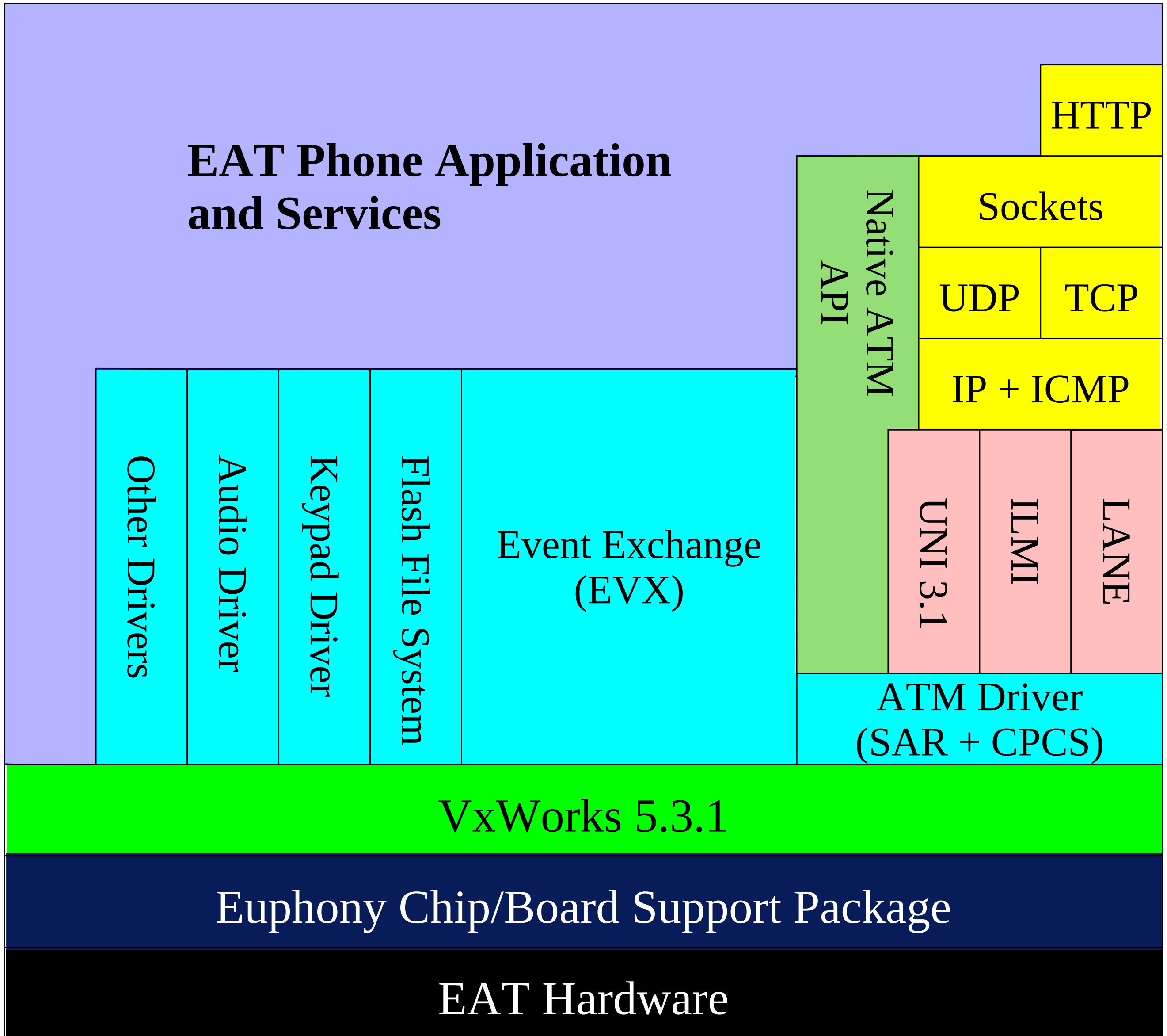
Euphony ATM Telephone (EAT) Block Diagram



EAT System Logic Board



EAT Software Architecture



Why an RTOS?

Why do you need an OS at all?

- Packet telephony application quite complex
- Need tasks and interrupt handlers
- Need standard libraries

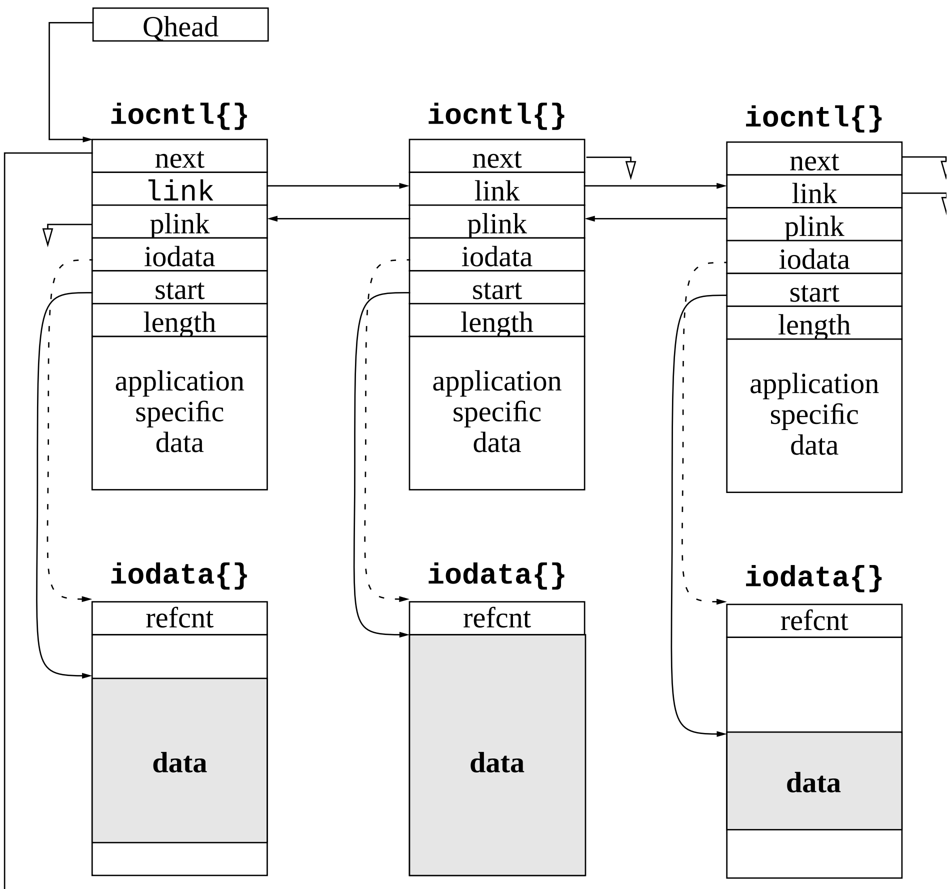
Why not Linux?

- Timesharing environment brings too much baggage
- Hard to scale down, Not real-time
- Poor embedded development tools

Why VxWorks (could be any RTOS)

- Small and efficient real-time kernel
- Scalable (minimum kernel ~64K-bytes)
- Ported software (web server, SNMP, Java, etc....)
- Excellent cross development tools

IObufs



- **Copy reduction techniques used to reduce memory & latency**
- **Similar to BSD Unix mbufs**
- ***IObufs* are not mbufs**
  - **Allows application specific info**
  - **Separate control and data blocks**
  - **Data block can be of any size**
- ***IObufs* provide a uniform buffering mechanism used by all modules**
  - **I/O (e.g., network and audio)**
  - **Application**

Intra-Appliance Communication

Initial software implementation was too unstructured

- Tightly integrated, hard to debug and add new features
- Motivation for a modular design that allows evolution and experimentation
- Avoid tight coupling between appliance functions
- Flexible communication for media buffers and events
- Example events: call states, on/off hook, key presses

Efficient audio movement using zero-copy IObufs

- Coupled with EVent eXchange (EVX) for distribution
- One-to-many, flow controlled, sender/receiver decoupling

Examples of EVX use

- Key presses sent to digit collector and tone generator
- Hook events potentially of interest to several modules

EVX

EVX delivers events posted on a “sending” port to one or more interested modules on their “receive” ports

- Module communication defined in terms of port names and event types
- Events can be delivered to multiple receivers
- Sender does not need to know about receivers
- Data delivered in IObufs using reference counts
- Flow control to prevent overrun

EVX API for creating ports, sending/receiving events, waiting for events and network I/O

EVX application consists of three parts

- Application modules
- Initialization code that configures EVX connections
- EVX API library + EVX thread for event processing

EAT Software

I/O

- Audio, network
- Keypad
- Hook monitor

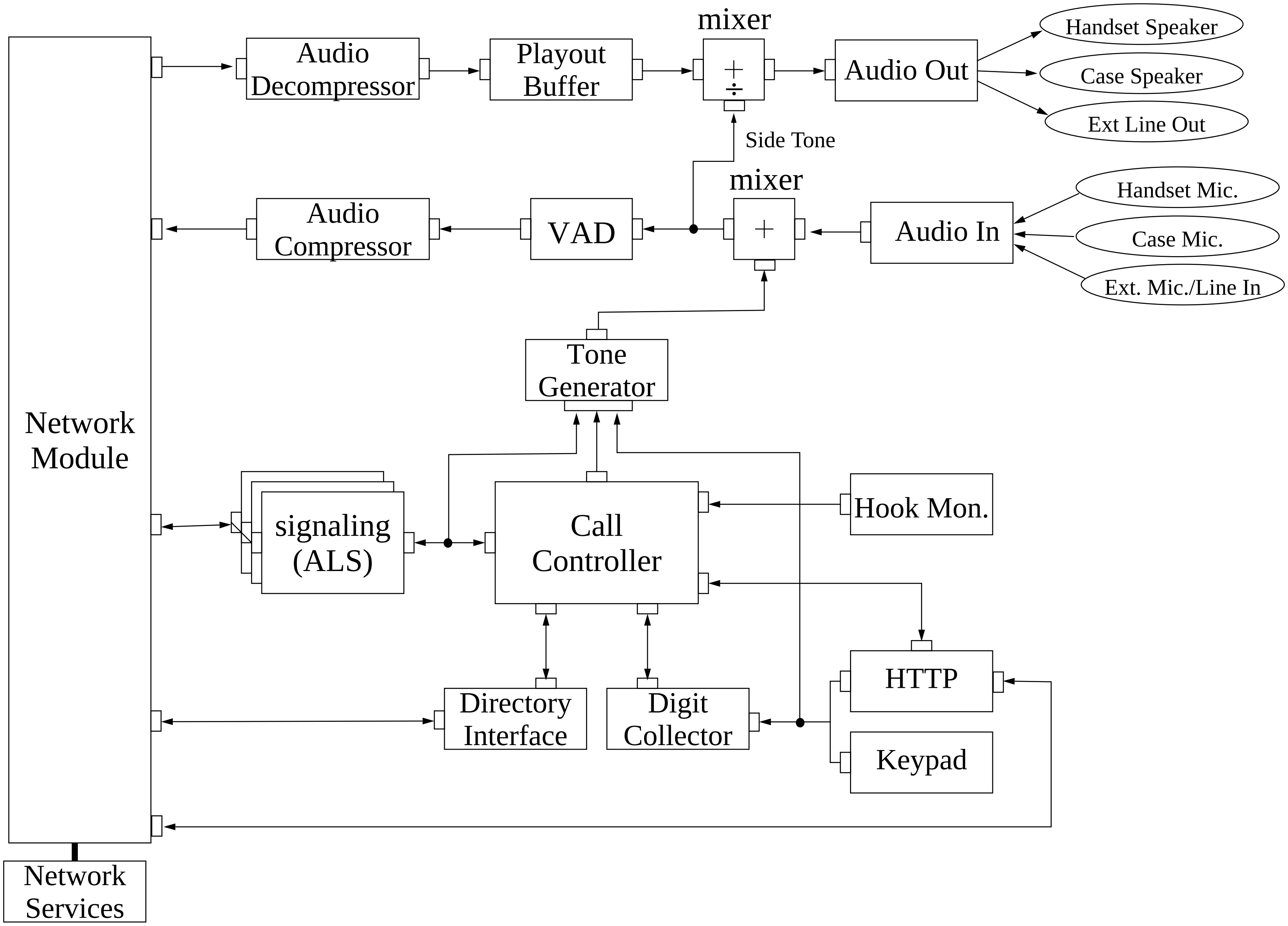
Audio path

- Compression/Decompression
- Voice Activity Detection (VAD), Playout buffering
- Mixing audio samples

Telephony

- Tone generator
- Signaling, Directory access, Call controller
- Digit collection

EAT Modules and Data Flow



Status and Future Work

Ongoing

- Deploying more than 20 EATs in offices
- Building a T-1/PBX gateway using ALS

EVX

- Dynamic configuration of new services, coders etc

Signaling

- Use of per-call choice of signaling protocol

Advanced Services

- What features should be in network servers and what features should be implemented by “intelligent” devices?
- High quality music end-point
- Voice enabled user interface
- Interactions with network services

Related Work

Packet Telephony Directories and Call Signaling

- IETF Session Initiation Protocol (SIP), ITU H.323, etc

Packet voice

- Early voice networking, e.g. Etherphone, ISLAND

IP Telephony appliances

- Off-the-shelf (e.g. Selsius, Symbol)
- Fixed choice of coding and signaling (H.323)
- Services via LAN-based PBX PC

Communication

- Zero-copy techniques, Fbufs, Rbufs, etc
- Distributed event services
- Conference bus protocols, Message/software buses

Conclusion

Comprehensive design and implementation of a packet telephony appliance

- Low-cost, simple telephony device
- Easy to use and reliable
- Suitable for experimentation

Hardware

- Networked processor with ATM and digital audio interfaces
- Traditional styling, 12-button keypad, handset

Software

- Real-Time OS, ATM & IP protocol stacks
- Efficient and extensible: zero-copy IObufs and EVX
- Advanced features and control using a web browser