Internet2 VoIP Conferencing Service

Internet2 VoIP Workshop
TAMU, College Station, Texas

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Motivation

PSTN bridging sucks

- Very expen$ive
- Not **our** dogfood
- Low fidelity (*i.e.* POTS)

Promote an interesting interactive application

- Push VoIP in an interesting direction (wideband)
- Internet2 conferencing efforts have attracted few users
- VoIP has potential to attract a huge user base

Ulterior motives

- Promote SIP
- VoIP is perfect "canary" for E2E−perf faults
- VoIP user base + functioning TCPs = QoS demand?

Concept

Deploy SIP−based conferencing solution internally and for Internet2 WG calls

Aim for better−than POTS fidelity

Gateway(s) to support legacy users

Incentives for users to switch

- **Carrot**: better−than−POTS fidelity, IP purity
- **Stick:** no more 800 access; users connecting from PSTN would pay their own long−distance

SIPConf$_1$

Based on Columbia/SIPComm's CINEMA

Runs on Solaris or Linux

SIP for signaling and RTP for media

Supports multiple simultaneous conferences

- Scales up to 100 users on current harware
- Work underway to allow a conference to span multiple servers

Codec heterogenity (through transcoding)

Playout delay algorithm to create a synchronised mix

SIPConf$_2$

Web−based administrative interface

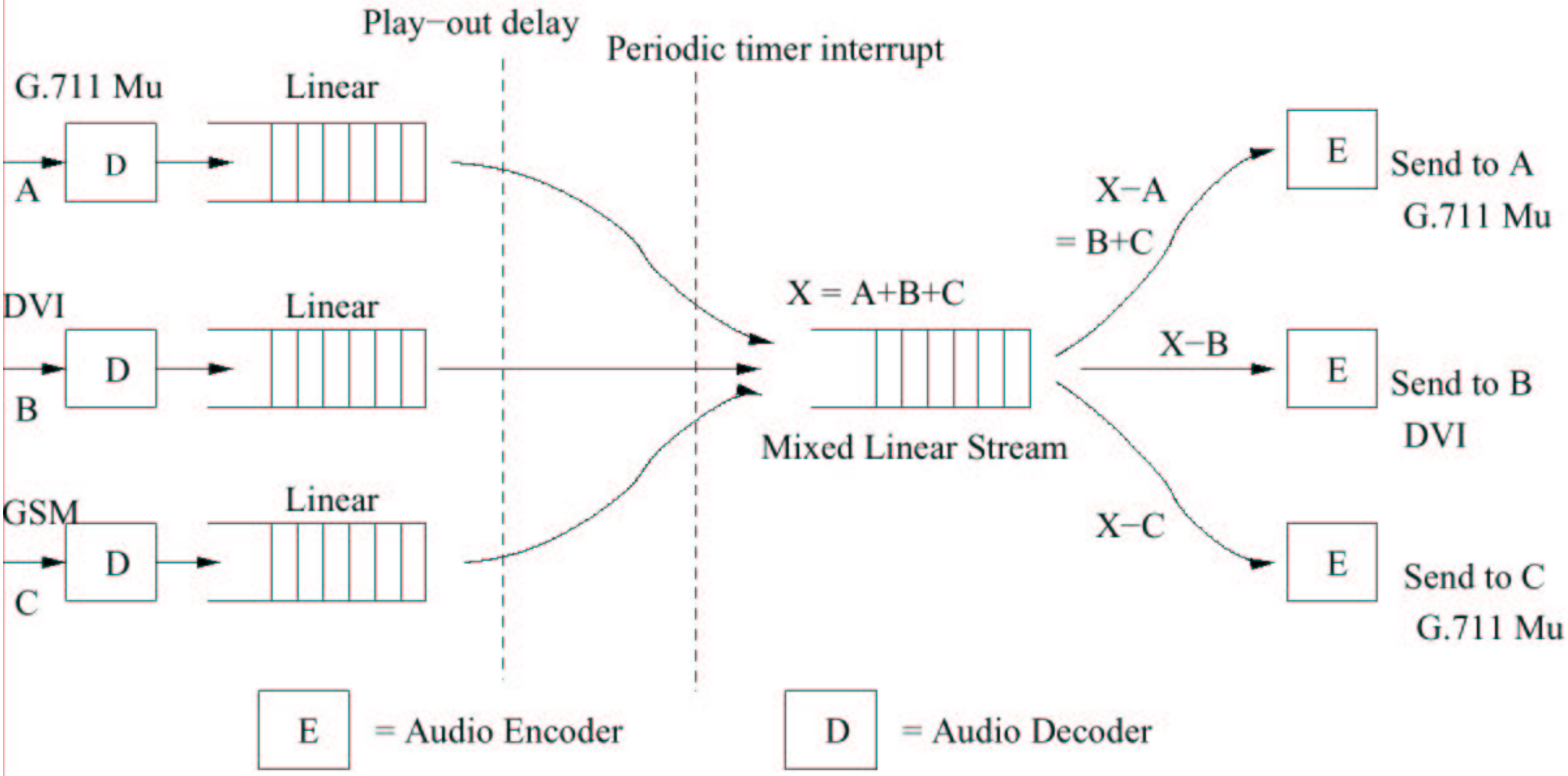
Currently supports G.711 A law, Mu Law, G.721 and DVI ADPCM audio codecs ***(now also G.722)***

IPv6 enabled (session control and media)

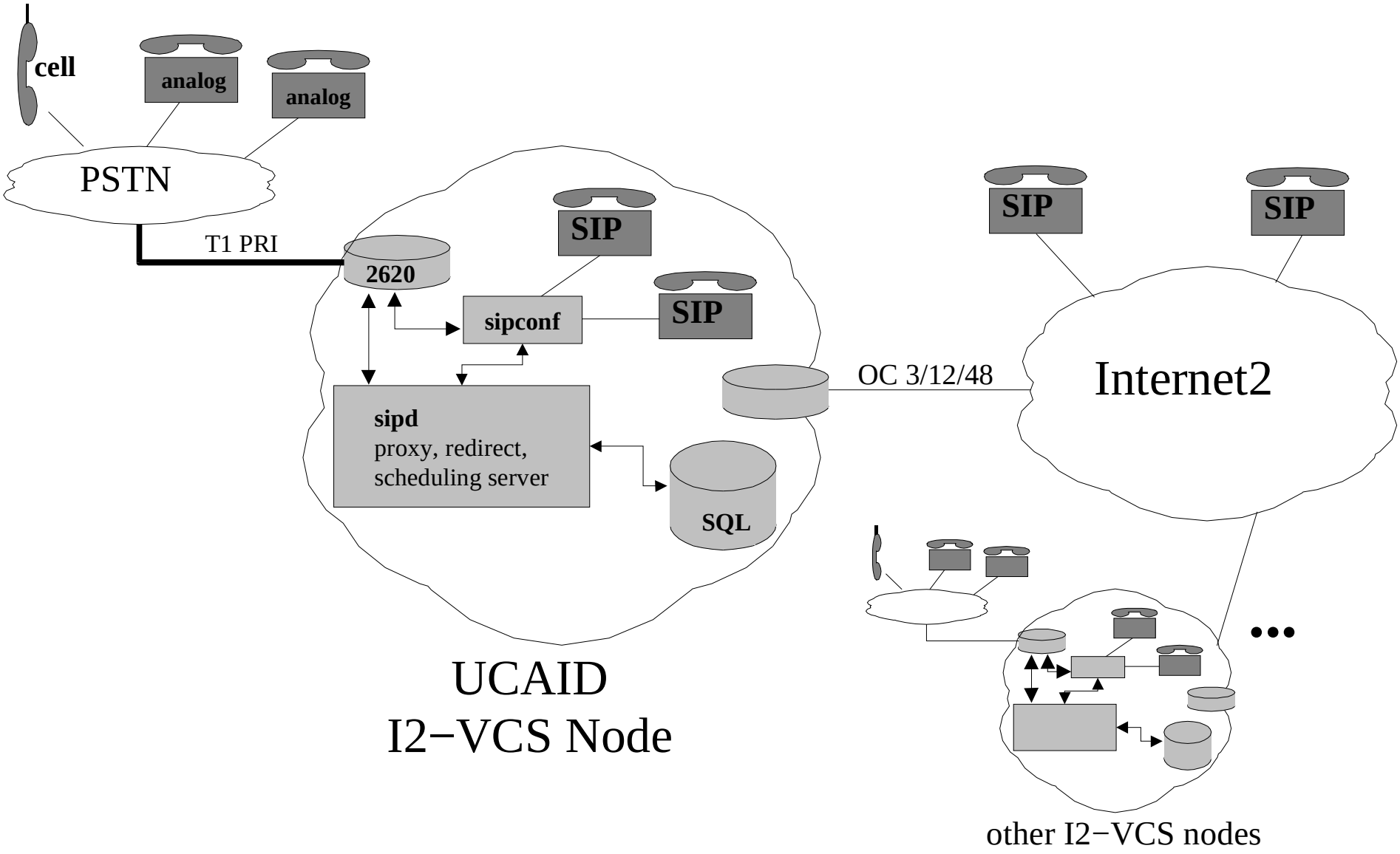
And more...

- File sharing
- IM
- Recording
- Video
- Whiteboard (through vnc)

Mixing and Transcoding



I2−VCS Architecture



Status

Testing in cooperation with Yale (many thanks to Jeremy George!)

Warts, yes, but it works

Next steps

- Working with Columbia to improve admin GUI
- Internal demo for senior management
- Early adopter internal weekly internal call
- Production use (I hope!)
- Discussion underway to support other wideband codecs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