Internet Real-Time Laboratory

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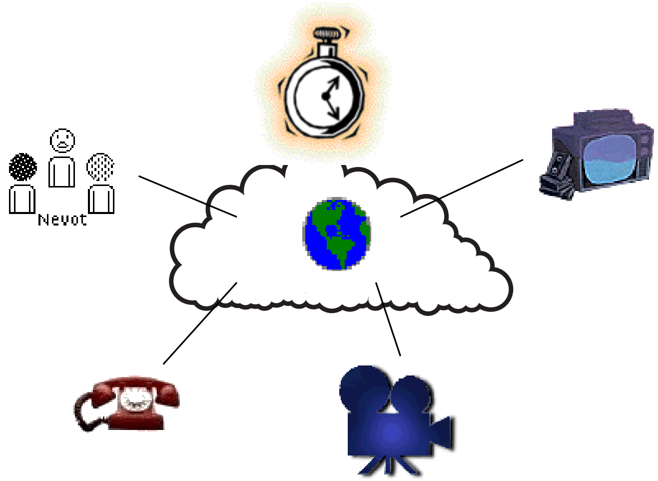


IRT Members: Full-Time PhD Students

Student	CU	IRT	research topic
Jonathan Lennox	F1994	S1997	call processing language
Wenyu Jiang	F1995	F1998	Internet QOS measurements
Xin Wang	F1996	F1996	multicast, QOS, pricing
Maria Papadopouli	F1996	F1997	ad-hoc networks
Fatima Al-Garawi	F1997	F1997	distance learning
Weibin Zhao	F1997	F1997	directory services

IRT Members: Part-Time PhD students

Student	employer	research topic
Lisa Amini	IBM	video on demand
Mandis Beigi	IBM	resource reservation
Ethen. Bommaiah	Bell Labs	media on demand
Ashutosh Dutta	Telcordia	MarconiNet
Christopher Kang	Bell Atlantic	signaling translation
Santosh Krishnan	Bell Labs	scheduling, QOS
Ping Pan	Bell Labs	resource reservation
Jonathan Rosenberg	Bell Labs	Internet telephony

IRT Members: Masters Students and Visitors

Gautam Nair		1999–	Internet telephony
Xiaotao Wu		1998–1999	SIP implementation
Jianqi Yin	BUPT, Beijing	1997–1999	Internet telephony
Elin Wedlund	Ericsson, Sweden	1998–1999	mobility
Tony Eyers	U. Wollongong	1999	signaling performance

Research Topics

- network infrastructure
  - scalable resource reservation protocols
  - QOS measurements and management
  - adaptive services and pricing
- Internet telephony (VoIP)
  - signaling: SIP, translation to ISUP/H.323/MGCP
  - VoIP services and service creation
  - e*phone
- applications
  - "civil society" computing
  - Internet TV and radio
  - distance learning

QoS Measurement and Management

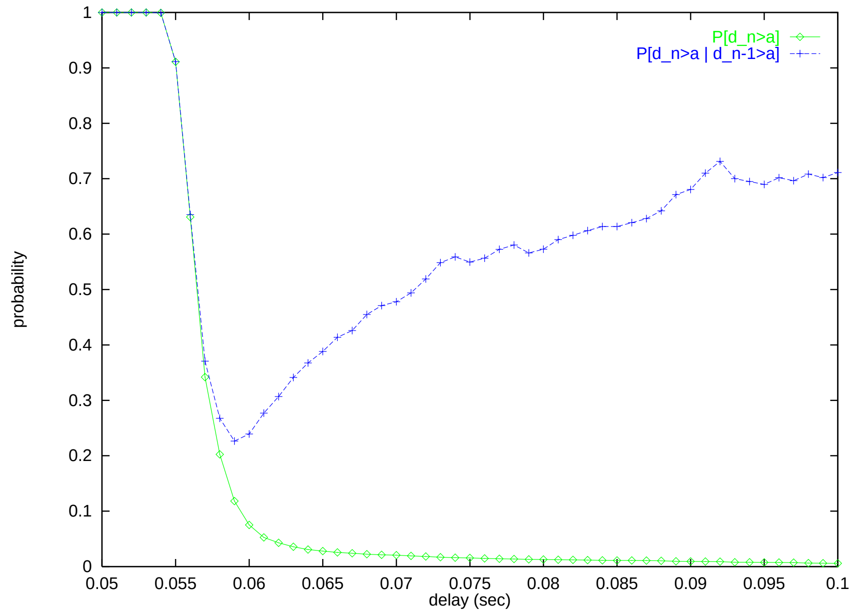
- packet delay and loss characteristics in the Internet
- inter-domain QoS Monitoring and fault location
- study of user behavior in IP telephony

One-way End-to-End Packet Delay Measurement

- Measurement without clock synchronization:
  - cannot precisely measure absolute value of delay
  - but sufficient for analyzing delay variation (jitter)
  - need to compensate for clock drift for long measurements
- Combining traditional telephone network and IP network:
  - reasonably precise in measuring absolute value of delay
  - requires special telephone-audio converters
- Using GPS as globally synchronized time source:
  - best precision
  - but requires special (expensive) GPS equipment

Modeling Packet Delay and Loss Correlation

- packet losses are correlated; can be described with n-state Gilbert model
- delay and loss are correlated:



Prediction of User-Perceived Quality

- short-term QoS statistics such as loss rate are essential to quality prediction.
- first-order statistics such as average loss rate do not capture user-perceived quality: e.g., burst losses vs. random losses
- FEC or low bit-rate redundancy, coupled with error concealment, further improves (and complicates) measure of user-perceived quality.
- signaling performance also needed

Inter-Domain QoS Monitoring and Fault Location

- SNMP statistics provide per-host information ➠ difficult to pinpoint the source of trouble in a large network
- use (extended) RTCP statistics for e2e quality monitoring
- hierarchy of monitoring stations
- aggregate across multimedia sessions

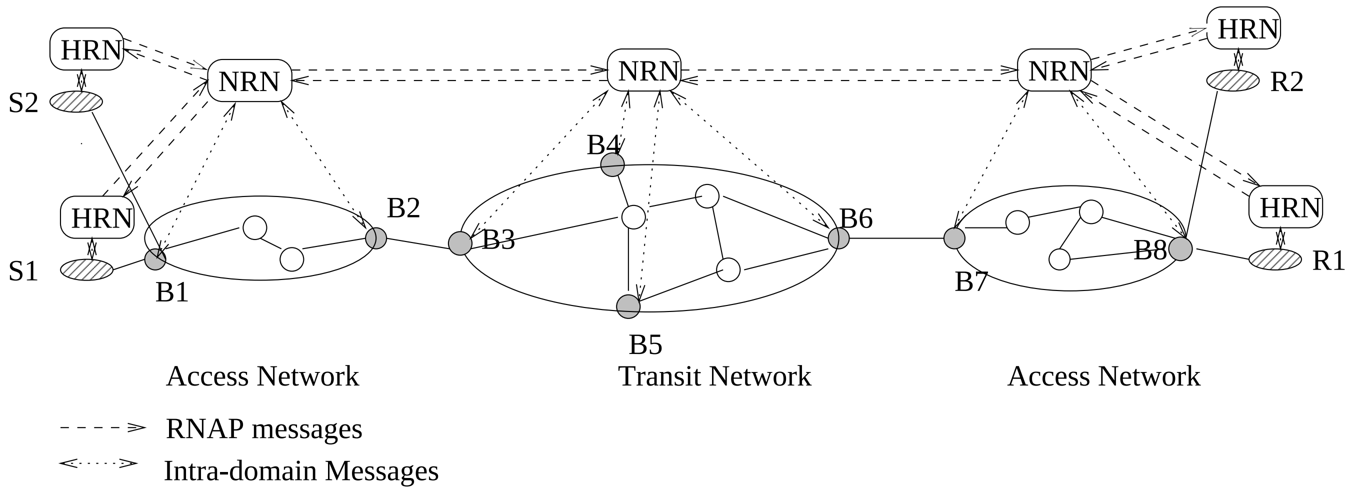
User Behavior in IP Telephony

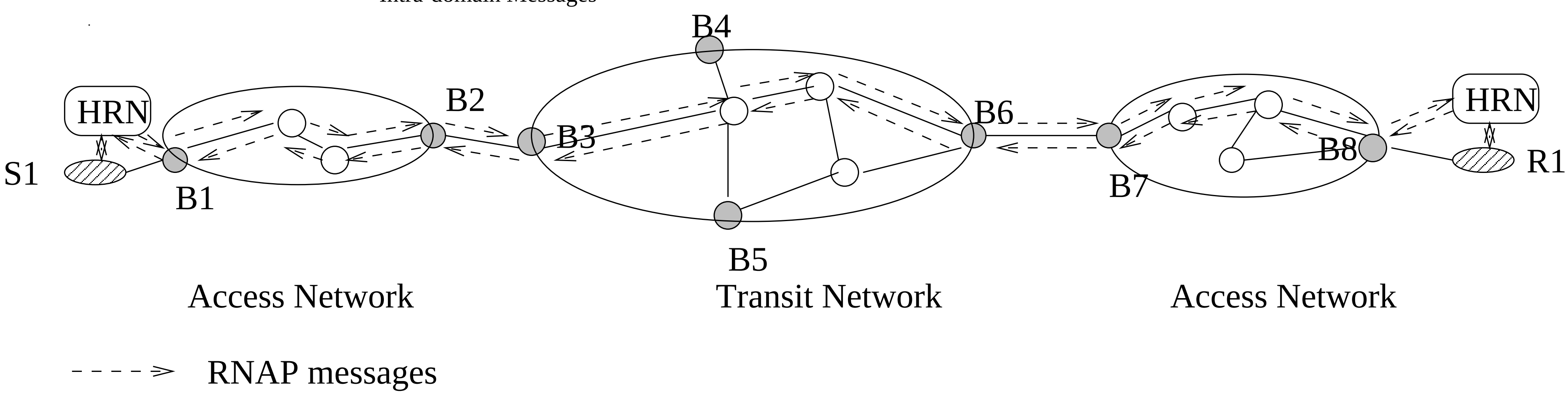
- will user behavior change from PSTN to IP Telephony?
- call attempt and retries?
- speech's on-off pattern
- feature usage: time-of-day forwarding, filtering, voice/email, ...
- user trade-offs: quality vs. cost
- ➠ test bed at Columbia University planned

Pricing Adaptive Multimedia Services

- pricing schemes that consider the holding cost, usage cost, and congestion cost as well as supports differential quality and advance reservation in the Internet.
- develop a network infrastructure that supports dynamic resource negotiation and pricing across Internet domains and users ➠ RNAP

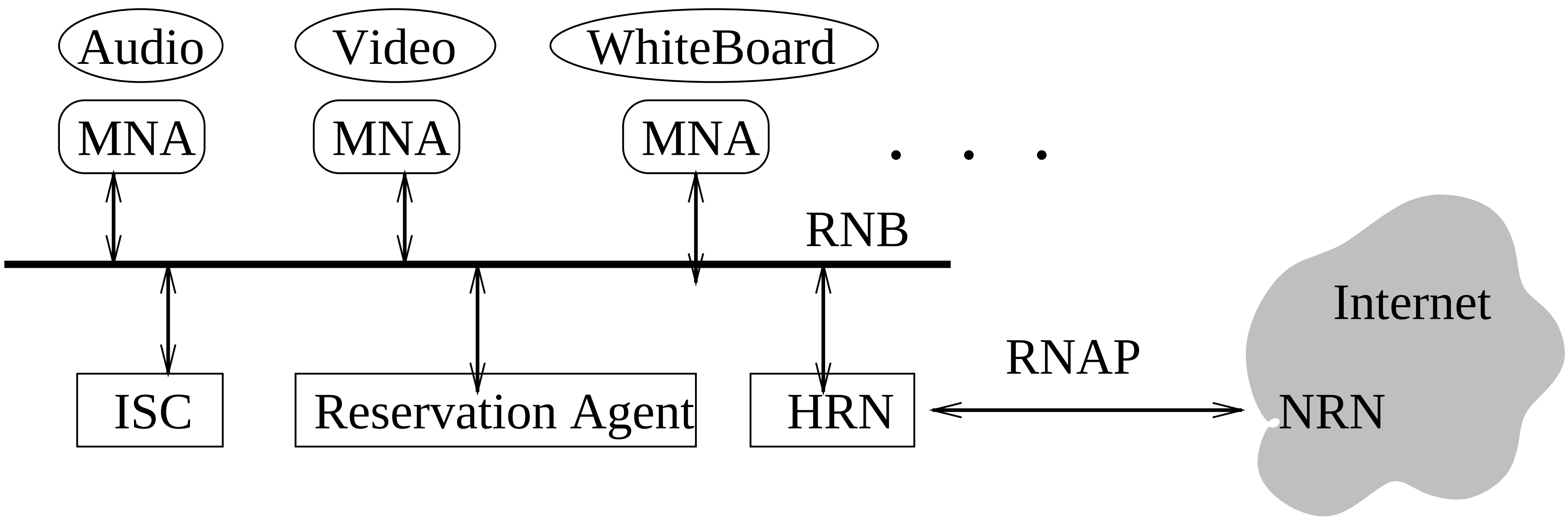
RNAP Architecture





Adaptive Multimedia Services

- design an incentive-compatible adaptation framework that optimizes user satisfication and network revenue, minimizes connection rejections.
- cross-media adaptation scheme to optimize system wide performance;
- designing test methods for user utility functions for different tasks



IP Multicast and Network Reliability

- performance analysis of different multicast protocols (DVMRP, MOSPF, CBT, PIM-DM, PIM-SM)
- failure tests of routing protocols PIM-DM and PIM-SM
- implementation of multicast protocols (PIM-DM, PIM-SM) in Opnet environment.

Internet Telephony

SIP development: developing the SIP specification and various extensions for supporting IP telephony: call control, QoS, and PSTN interoperation.

SIP implementation: implementing a SIP server: full proxy, redirect, and registrar, with support for policy, and programmability.

IP telephony service creation: SIP CGI for service creation; implement Call Processing Language (CPL)

Internet Telephony

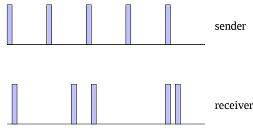
FEC: integration of FEC and adaptive playout buffer algorithms.

RTP: sampling algorithms for group membership tables in RTP; conformance tests for verifying RTCP implementation correctness.

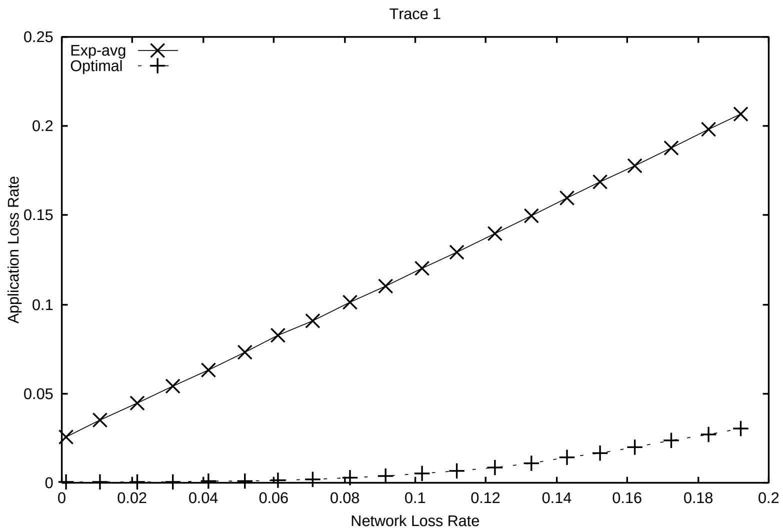
Presence: protocols and architectures (based on SIP) for presence and instant messaging.

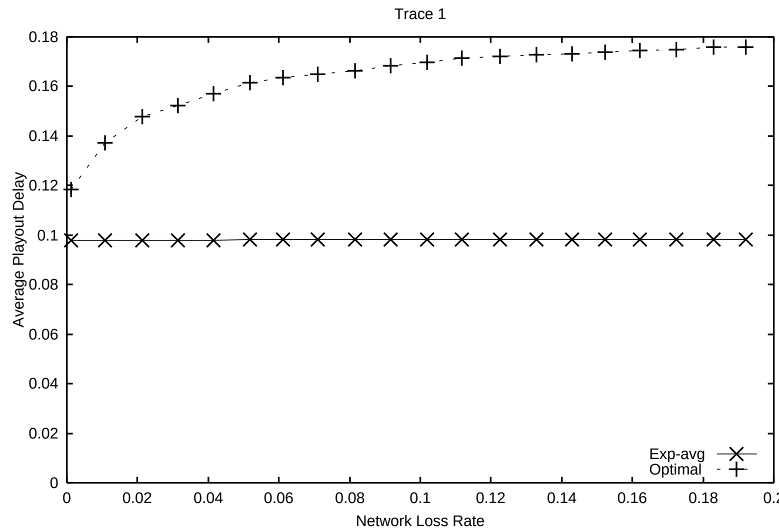
Integrating packet FEC into adaptive voice playout buffers

- playout buffer: trade loss (2…20%) for delay (50…500 ms)



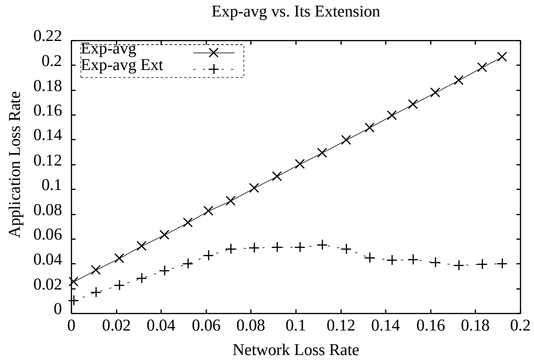
- (n, k) FEC: add $n - k$ additional packets for total of n

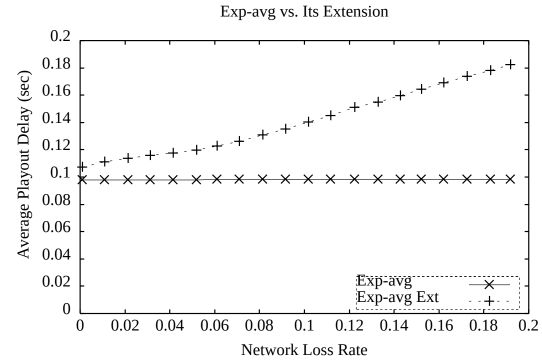


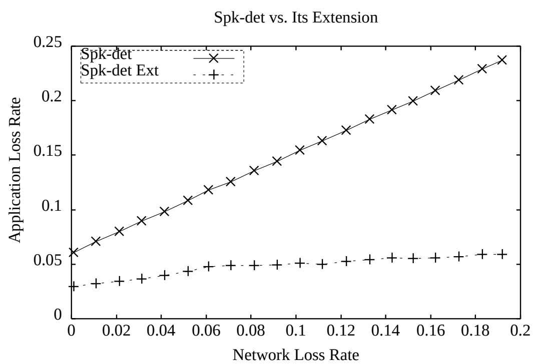


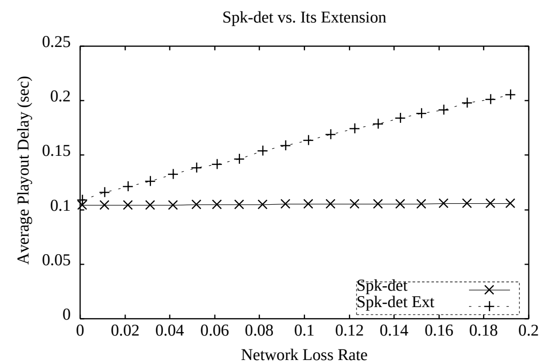
FEC: Virtual Delay

- virtual delay = min(arrival, recovery) $-$ departure
- playout delay $\approx \alpha \cdot \sigma$
- if loss $<$ target loss, $\alpha \leftarrow \alpha + \delta$
- recover lost and **late** packets
- 20% loss: application loss/5, delay * 2


Exp-avg vs. Its Extension
Application Loss Rate
Network Loss Rate
Exp-avg
Exp-avg Ext


Exp-avg vs. Its Extension
Average Playout Delay (sec)
Network Loss Rate
Exp-avg
Exp-avg Ext


Spk-det vs. Its Extension
Application Loss Rate
Network Loss Rate
Spk-det
Spk-det Ext


Spk-det vs. Its Extension
Average Playout Delay (sec)
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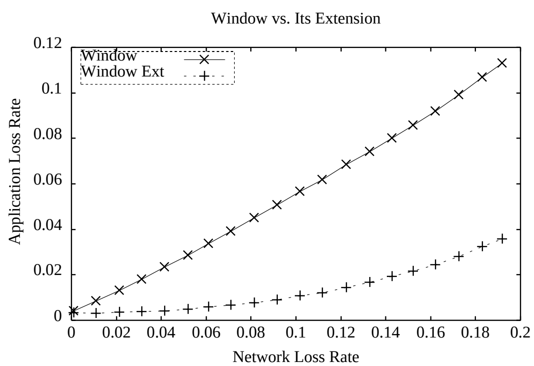
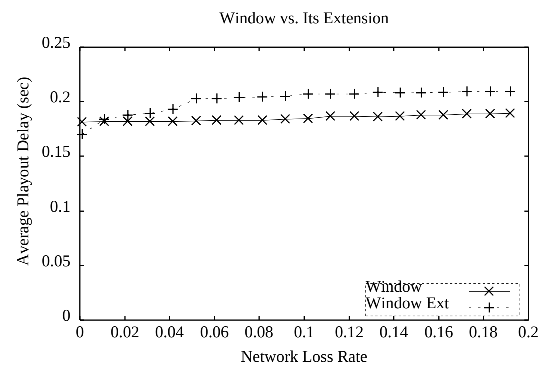

Window vs. Its Extension
Application Loss Rate
Network Loss Rate
Window
Window Ext


Window vs. Its Extension
Average Playout Delay (sec)
Network Loss Rate
Window
Window Ext

New Playout Algorithm: Delayed Optimal

- after talkspurt, one knows optimal delay
- use optimal combination for next talkspurt
- $D_i = \alpha D_{i-1} + (1 - \alpha) D_{\mathrm{opt}}$
- user perception function: minimal delay that achieves loss target

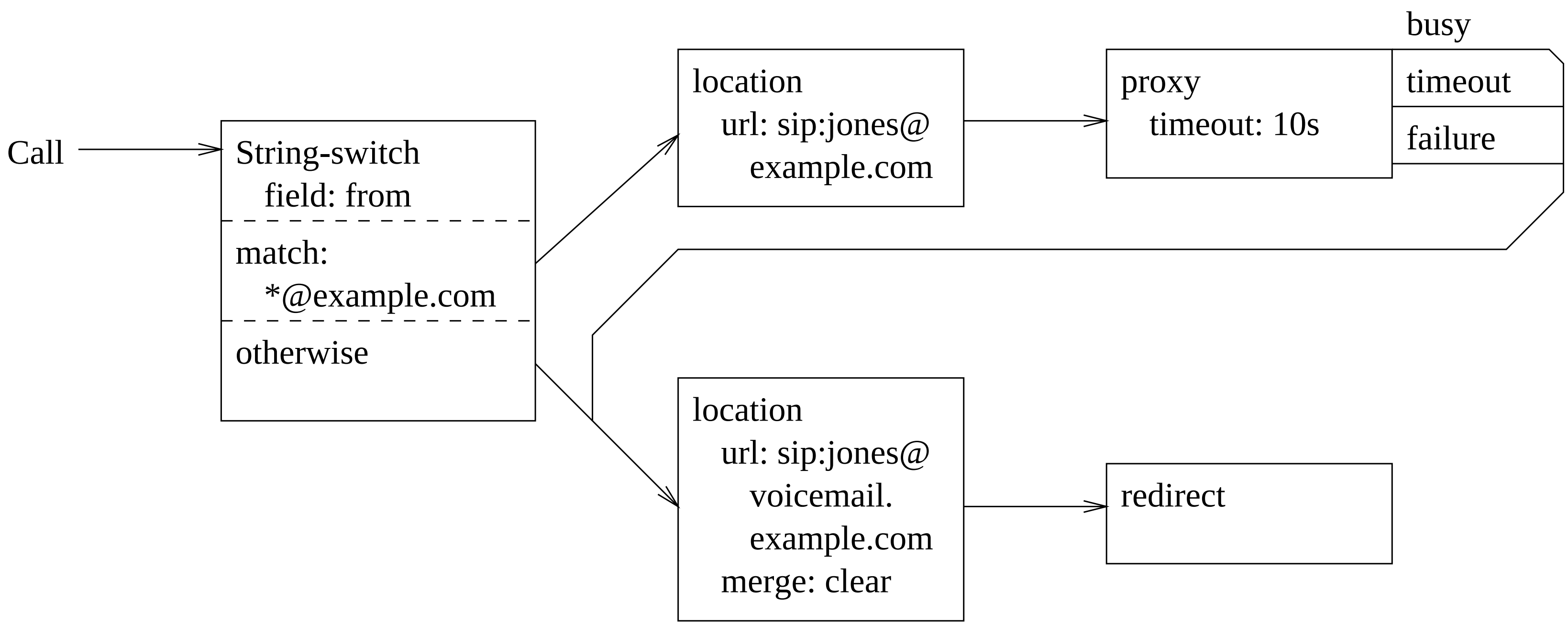
New Playout Algorithm: Analytical

- assume independent loss p, delayed randomly d
- compute playout probability
- tabulate delay distribution histogram
- at end of talkspurt, find matching playout delay for target loss rate

Call Processing Language for Internet Telephony

- enable Internet telephony features:
  - call forwarding
  - user location
  - call filtering
  - and many other features . . .
- let users control and customize features
- help from provider not needed
- simple, safe scripting language

Call Processing Language



- no loops or variables ➠ safe execution
- easily created in graphical environment
- XML-based ➠ easy to parse

RTP and SIP Implementations

- Columbia/Elemedia RTP library
- Columbia SIP proxy/redirect/registration server sipd
  - user registration
  - forking proxy capability: one call, many destinations
  - sip-cgi: interface to scripting languages
  - security mechanisms for authentication
  - caller preferences

E*phone: An Internet Phone Appliance

- phone = $49.99; PC $>$ $600 (GPF included)
- *Ethernet phone* ➠ no PBX for switching
- DSP for voice coding ➠ limited memory (512 kB SRAM!)
- minimal real-time OS (threads, signals)
- implemented minimal IP stack (IP/UDP/RTP, DHCP, SIP)
- no TCP, no DNS
- MP3 radio?
- interface to the real world: alarms, measurements, ...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VoIP Signaling Translation and Gateways

H.323 Decoder

- identify the packets that contain signaling informations in Q.931, H.225.0 and H.245;
- decode the signaling packets coded in Packed Encoding Rules (PER) X.691

SIP, ISUP and H.323 Gateways

SIP $\leftrightarrow$ H.323 gateway:

1. identify SIP requests and headers for H.323 signaling messages;
2. develop of SIP in-call control messages if necessary;
3. build SIP–H.323 gateway

SS7 $\leftrightarrow$ SIP gateway:

- ISUP-to-SIP handoff & SIP bridging
- map SIP requests $\longrightarrow$ SS7 ISDN User Part (ISUP)
- implement prototype gateway

Social Mobile Computing

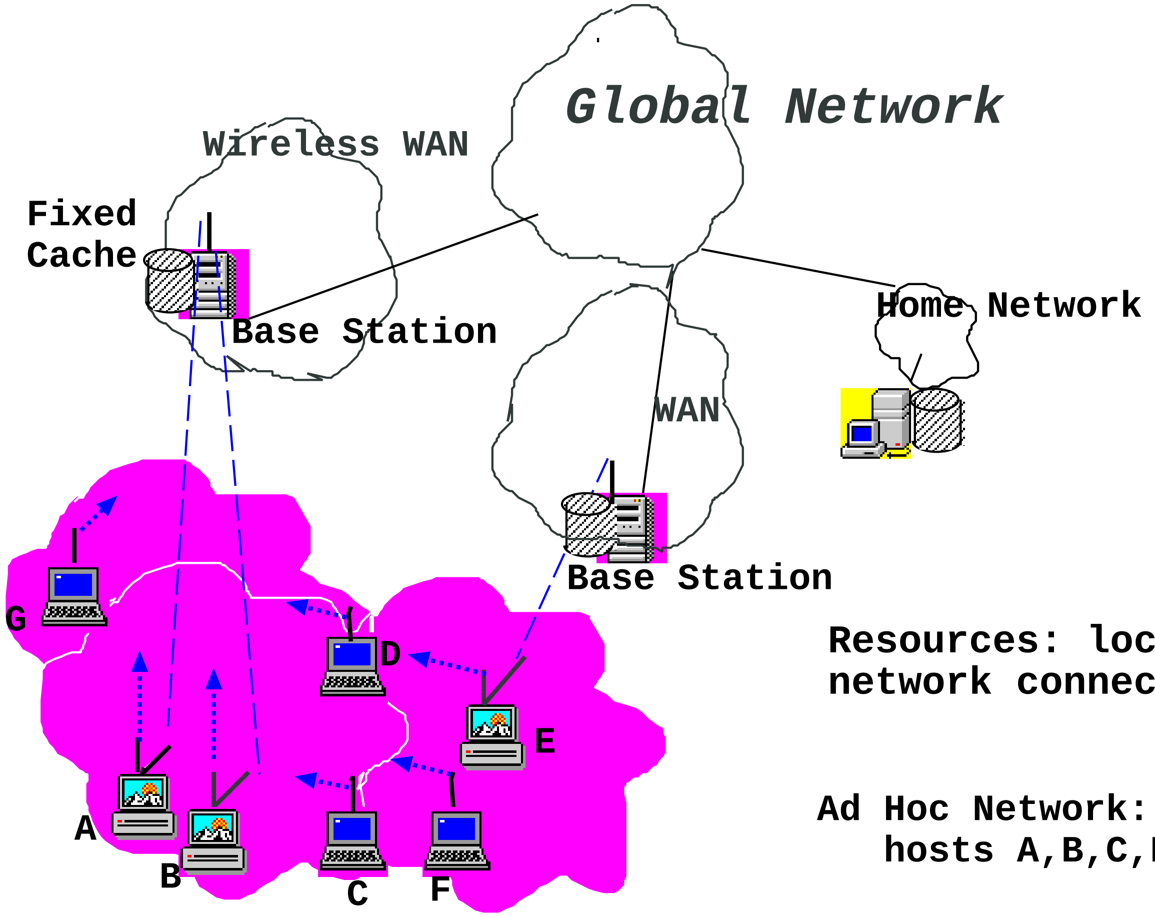
- provision of multimedia applications to mobile wireless users:
- ad-hoc mobile networks combining different wireless technologies;
- collaborative multimedia applications: teleconferencing, web browsing, sharing notes, multicast discussion
- anticipatory caching and prefetching

Ad-hoc Wireless Networks

- Users wish to collaborate by creating a wireless network “on the spot”, dynamically ➠ teleconferencing, news on demand, e-white boards,
- Current wireless deployment:
  - limited wireless network access to global network,
  - intermittent connectivity,
  - low bit rates, and
  - high end-to-end delays

$\Rightarrow$ *better utilization* of the user’s local resources to achieve higher *data availability* and *QoS*.

Resource Sharing of Collaborating Hosts in Wireless Adhoc Network



Resources: local cache, network connection

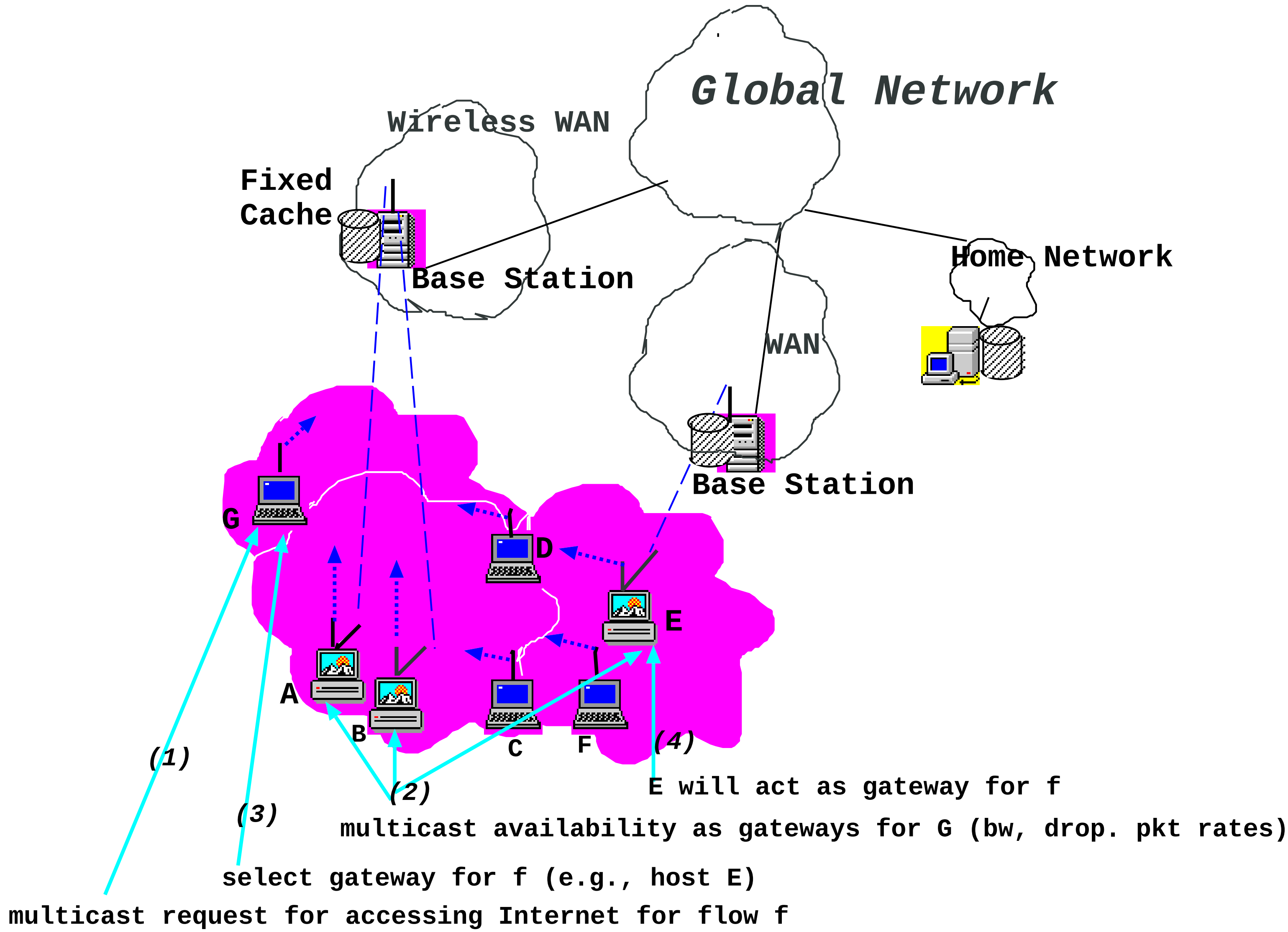
Ad Hoc Network:
hosts A,B,C,D,E,F,G

Temporal Gateways:
(dual home hosts) A,B,E

Motivations for Resource Sharing

- reduction in the transmission of replicated data that belong to “shared” (collaborative) applications.
- utilization of temporarily idle connections
- increase QoS for the shared-data applications
- financial benefits (e.g., resource renting, bandwidth “co-op”)

Connection Sharing of Collaborating Hosts in Wireless Adhoc Network



Design Properties for the Resource Sharing Protocol

- sharing takes place “on the spot”, dynamically $\Rightarrow$ lightweight protocol (low complexity and overhead)
- reduction of the replicated data that users are interested in viewing
- host selects host-gateway for a flow based on the gateway’s bandwidth availability and packet loss rates
- simply select *least loaded* gateway ➠ load balancing
- lightweight and secure micro-payment method needed
- need security and privacy for different types of collaboration

Multimedia on Demand Servers

- heterogeneous network environment
- admission control, scheduling and QoS guarantees
- server fault-tolerance and reliability issues
- storage issues: memory hierarchy and data layout

MarconiNet – Internet Radio & TV

- large and small audiences
- services: automatic insertion of local contents, advertising, media-on-demand services, switching between global and local program
- anybody can be a potential broadcaster
- access to any station in the world
- support for heterogeneous receivers in the local domain
- pricing structure for the network affiliates
- hardware dedicated to Internet radio/TV?

MarconiNet

- hierarchically scoped multicast for scalability
- components: primary station, local station, Internet clients
- RTCP to trigger automatic insertion of local contents, commercials
- RTSP server for media-on-demand application
- popularity-based spectrum management for bandwidth control
- extension of SAP/SDR to provide scalable directory structure
- novel program management at the local station
- secured distribution of paid programming
- Java-based user interface

Other Research

- session control architecture for the next-generation network
- IP/PPP/SONET, IP/ATM/SONET and IP/WDM issues
- power management for low-earth orbit satellites under bursty broadband traffic

Interactive Distance Learning

- CLIC lab + classroom:
  - 40 workstations
  - analog TV production facilities: remote-control cameras, video mixers, wireless microphones, . . .
  - designed for off-site participation
- experimented with existing video-conferencing tools (vic/vat, rendezvous, IP/TV) ➠ integrate to
  - view lectures off-line
  - participate during class
- minimal post-processing of lectures
- cross-platform support
- variety of networks: modem, ADSL, Internet2

Distance Learning

- develop system to integrate video, slides, web pages and electronic whiteboard:
  - capturing of video, slides and whiteboard;
  - synchronizing audio and video with slides;
  - annotation;
  - interface to electronic whiteboard.
- “query” mechanism: off-campus students interact in real time with instructor.

Service Location

Replicated Server Placement

- measure network distance, bandwidth, delay
- use BGP4 to collect network distance metric (hop counts)
- combine distance metric with DNS for replicated server selection

Project MarketNet (DARPA)

- use pricing for access and resource control
- need price advertising and real-time updating
- location or path-dependent service pricing scheme
- ➠ Service Location Protocol:
  - given service name and some attributes description, provide related servers location
  - local area network service location discovery
  - wide area network service location discovery
  - implementation for SLPv2 completed