



Implementation of VoIP over Satellite Links

Ulla Birnbacher, Ronald Toegl

Institute of Communication Networks and Satellite Communications

Graz University of Technology, Austria

ulla.birnbacher@tugraz.at

Presented at IWSSC'05 in Siena (8th September 2005)

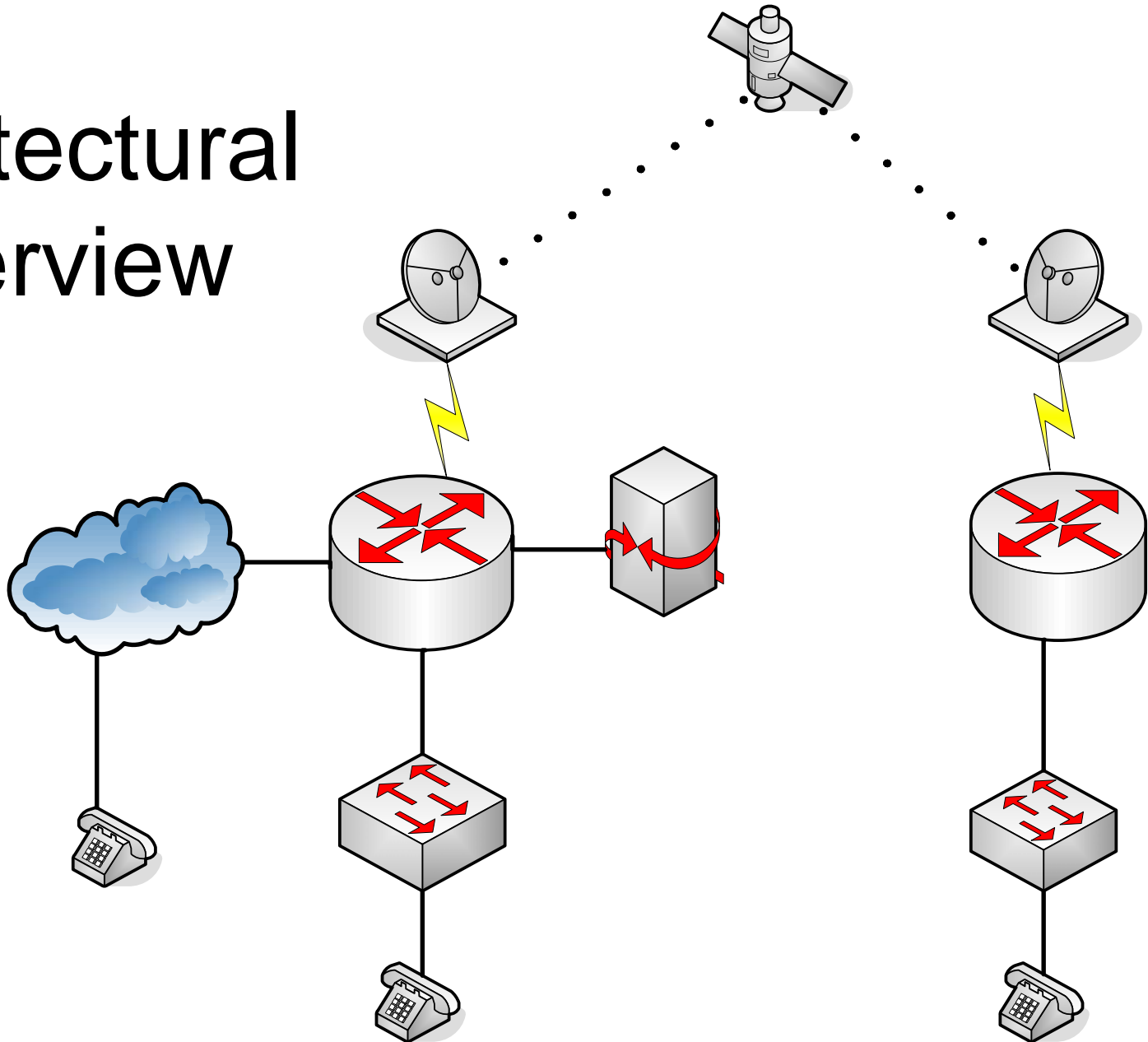
Outline

- Motivation
- Network architecture
- Implementation
- Results

Motivation

- All-IP Network over Satellite
 - Transport of any type of IP service
 - Video-conferencing
 - Voice over IP (VoIP)
 - Wide-area broadband connections
- Implementation of VoIP with the use of
 - Off-the-shelf network components
 - available QoS mechanisms

Architectural Overview



Satellite Communication



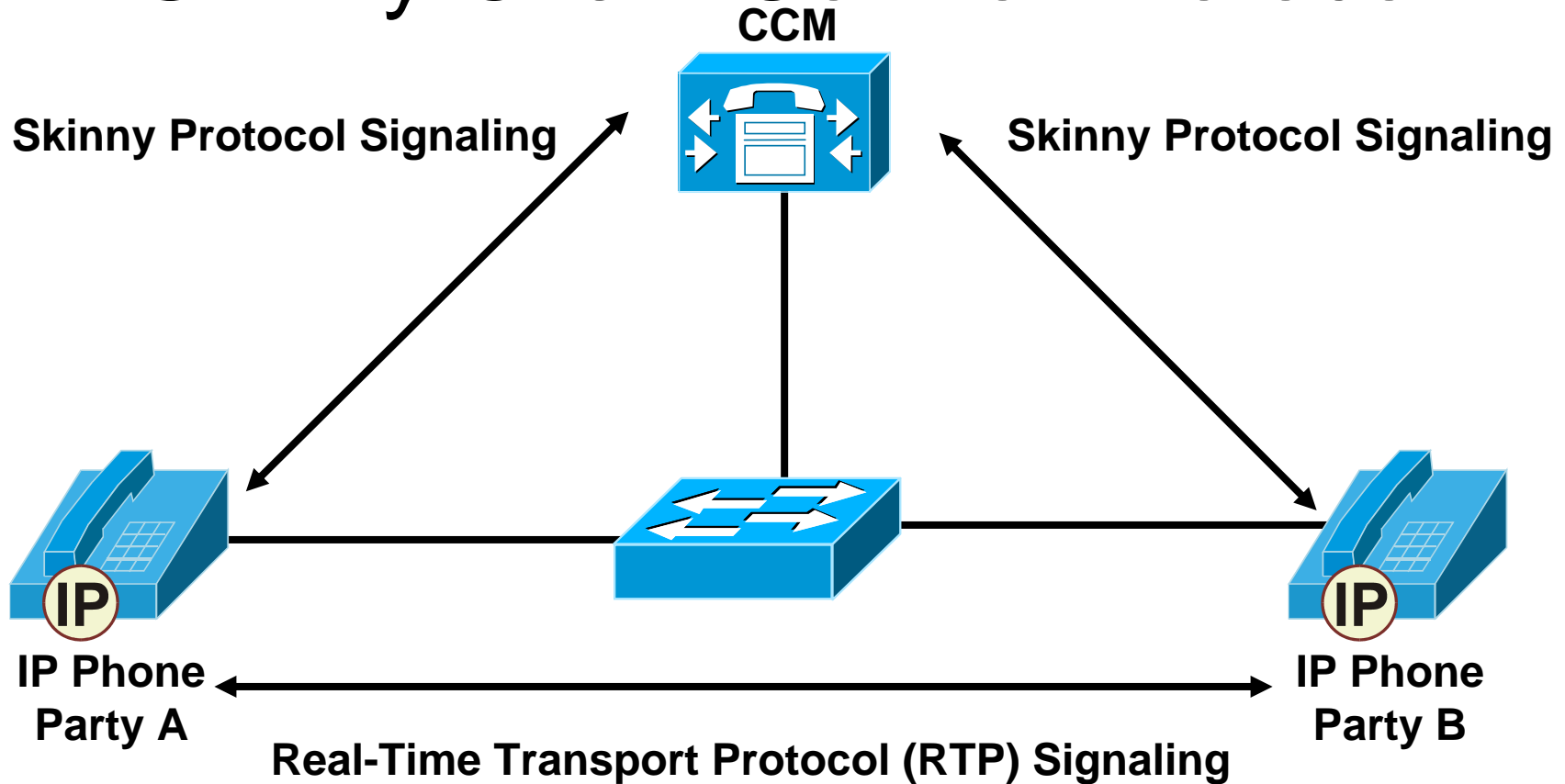
- Two 1 Mbit/s satellite links (FDM)
- Satellite: Eutelsat SESAT (36°E)
- Ground stations:
 - Parabolic dish antenna (1.5m and 2.4m)
 - Radyne Comstream Satellite Modem (QPSK, Viterbi FEC CR $\frac{3}{4}$)
 - Cisco Routers directly connected to modems by RS449

Cisco VoIP Approach

- Simple Clients
- Powerful Gatekeeper – CallManager
- Call control:
 - Cisco proprietary SCCP protocol
 - H.323 protocols possible
- Digitalized voice via IP/UDP/RTP/PCM

- When integrated in Router:
CallManager Express, for smaller setups

Skiny Client Control Protocol



- **Cisco CallManager performs call setup and maintenance functions.**
- **IP Phones stream audio using RTP streams.**

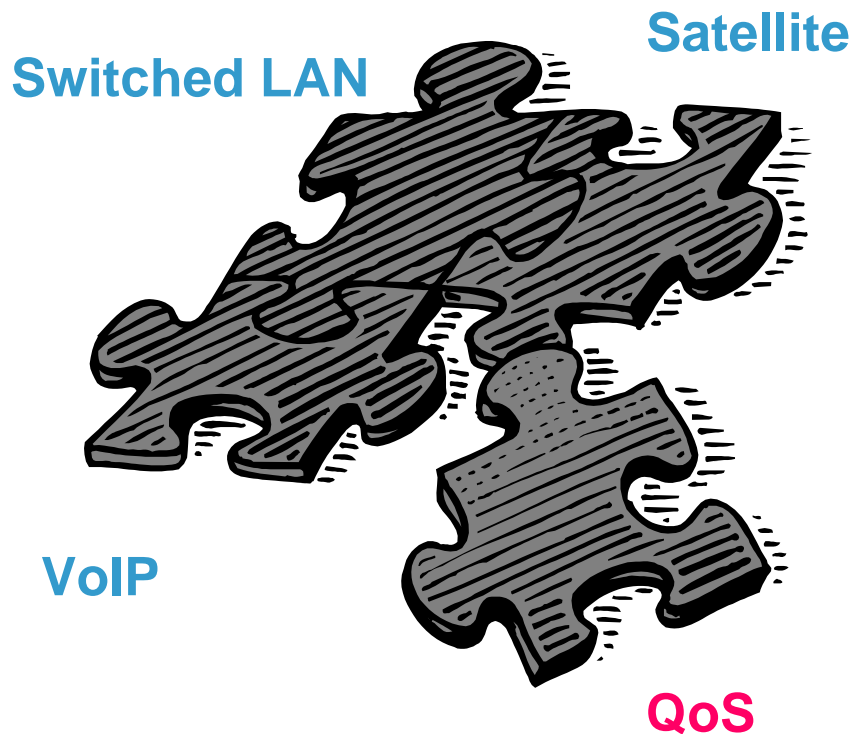
[J. Cioara. Introduction to Voice Over IP. Online video training, cisco.com, 2002.]

Implementation Hardware

- Cisco 1760V/2801 Routers with CallManager Express 3.0
 - Gatekeeper
 - ISDN-Gateway
 - NAT to Internet
 - Satellite Link
- Catalyst Switches
 - VLAN
 - Traffic Classification
- IP Phones



The Missing Piece



- Voice traffic is fragile
- Switched networks reduce some (local) problems
- It is easy to congest a 1-Mbit/s satellite link e.g. with ftp or video conferencing traffic
- **Quality of Service** guarantees needed

QoS Requirements for VoIP

- Packet Loss < 1%
- One-way, end-to-end Delay < 150ms
- Average Jitter < 30ms
- Guaranteed priority bandwidth
 - codec, packetisation interval
 - 27-93 kbit/s per call on Ethernet

[T. Szigeti and Ch. Hattingh. End-to-End QoS Network Design. Cisco Press, 2005.]

Differentiated Services Architecture

- Each packet carries **markings**
- Each packet class is treated **differently**
- Packet classification at DiffServ domain boundary only
- Each node has a **Per-Hop-Behavior (PHB)**

Per-Hop-Behavior

- Dominated by Packet Scheduling
 - Queuing
 - Scheduling
- Suggestions:
 - Assured Forwarding (RFC 2597),
 - Expedited Forwarding (RFC 3246)
- Still needs to be hand-tailored for each network and scenario

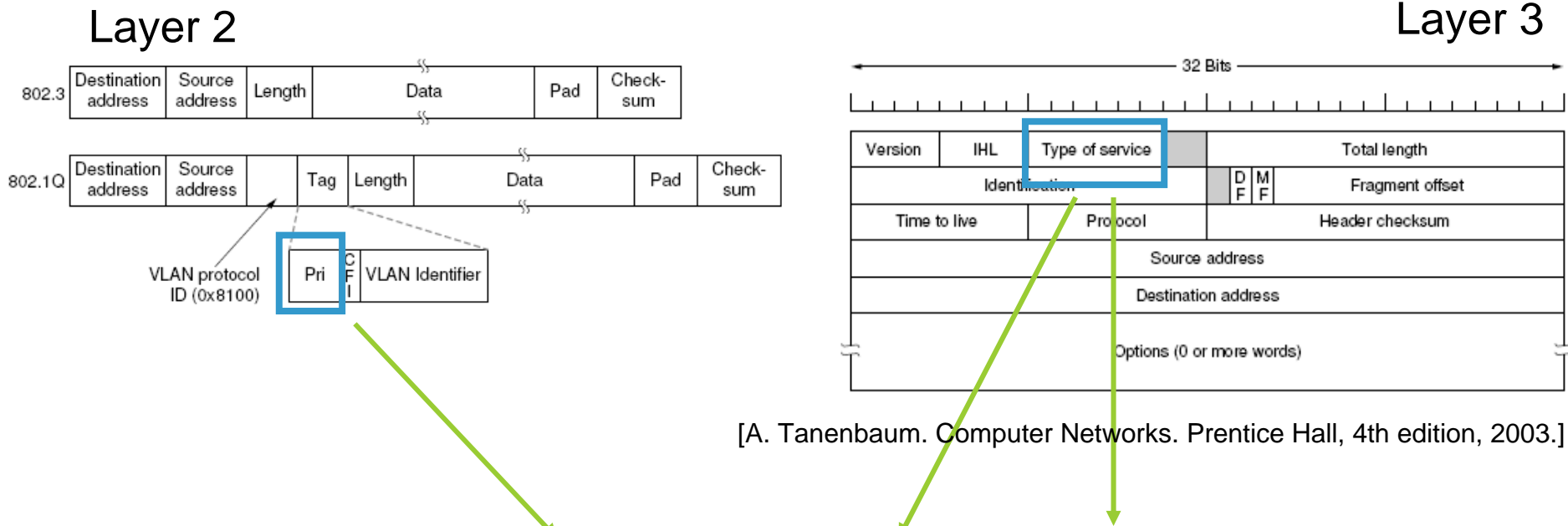
Designing DiffServ Networks I

- Map your applications to traffic classes.
“voice real-time”, “voice control”,
“video real-time”, “data”,...
- Define a class to marking mapping.

```
45 class-map match-all VOICE-CONTROL
46   match ip dscp af31
47 class-map match-all VIDEO
48   match ip precedence 4
49 class-map match-all VOICE
50   match ip dscp ef
```

(Cisco IOS)

Service Class Markings & Mappings



[A. Tanenbaum. Computer Networks. Prentice Hall, 4th edition, 2003.]

Traffic Type	Layer 2 CoS	IP Precedence	DSCP	PHB
Voice RTP	5	5	46	EF
Voice control	3	3	24	CS3
Video conferencing	4	4	34	AF41
Data	0, 1, 2	0, 1, 2	10 to 22	BE to AF23

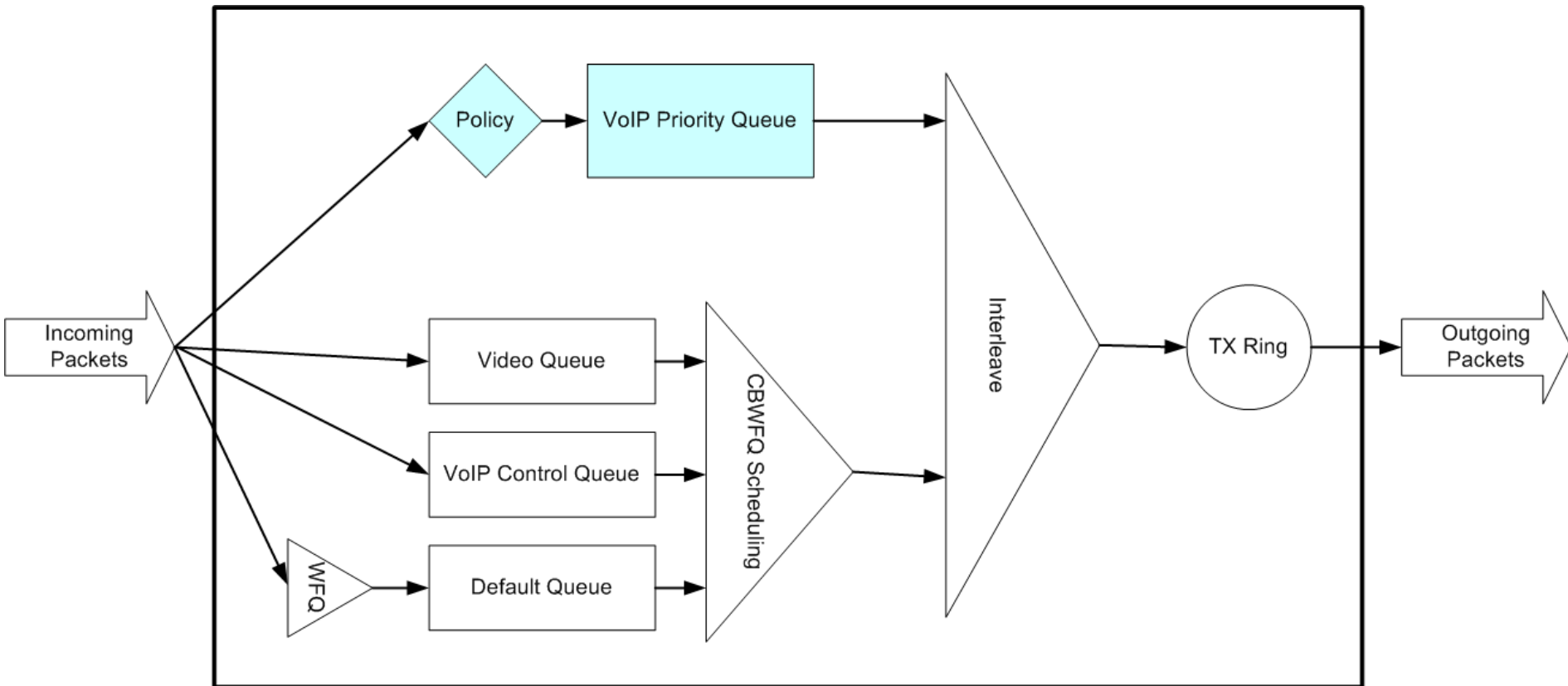
Designing DiffServ Networks II

- Setting of correct markings at domain boundary
- Configuration of the network behavior for each class in each node

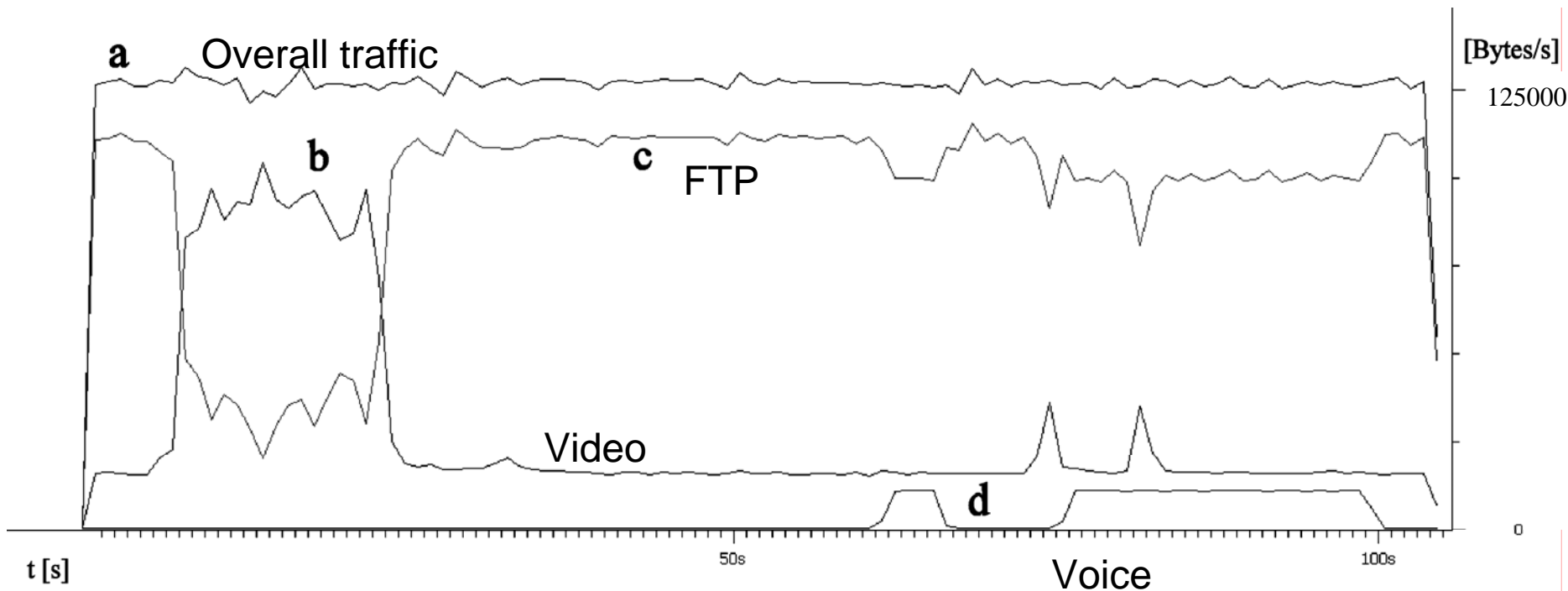
```
53 policy-map LAN-EDGE
54   class VOICE
55     set cos 5
56   class VOICE-CONTROL
57     set cos 3
58   class VIDEO
59     set cos 4
```

```
60 policy-map WAN-EDGE
61   class VOICE
62     priority 160
63   class VOICE-CONTROL
64     bandwidth 8
65   class VIDEO
66     bandwidth 740
67   class class-default
68     fair-queue
```

Low Latency Queuing for VoIP



Voice QoS in Action



Traffic type	Characteristic	QoS Settings
Voice (d)	64 kbit/s	DSCP 46, "Expedited Forwarding"
Video (b)	768 kbit/s	IP Precedence 4
Ftp-Data (c)	1 Mbit/s	DSCP 0, "best effort"
Voice control		DSCP 24, "CS3"

Results/Conclusion

- QoS mechanisms work
- QoS requirements for VoIP were met

Jitter	12.48ms
Reliability	2 in 2570 voice packets lost
One-way delay	42.55ms on ground plus propagation delay (~250ms)

- Perceived voice quality is very good
- Satellite link is used efficiently