QoS of Video Streaming over Mobile Internet

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1

1. Why networked video?

- video kids and N-generation
  - In 1969, Internet in USA
  - In 1976, color TV in Korea
- Reonardo da Vinci 21C
  - Art+technology

technology push → **Networked Video** → **market pull**

2

2. Classes of networked video service

- download
- Streaming
- Conversational (live)

❖서비스의 종류에 따라 요구되는 QoS가 다르다.

3

2. Classes of networked video service

2.1 download : replay after downloading encoded video

- 고속망을 통하여 짧은 시간에 압축된 비디오를 download받고, 자신의 컴퓨터에 저장하여, 자기가 편한 시간에 비디오를 시청하는 형태이다. 아침에 예약을 하고 저녁에 와서 시청하는 경우를 생각할 수 있다.
- 장점 : 통신망이 바쁘지 않을 때, download를 받는다면, 통신비용을 절약할 수 있다. 또한, 시간에 제약이 적어서 에러가 발생하면 재전송도 가능하므로 깨끗한 품질이 보장된다.
- 단점: 사용자 입장에서 단점은 즉각성이 없는 것이다. 서비스 제공자 입장에서는 비디오 콘텐츠가 복사되는 것을 막을 수 없으므로, 수익을 보장하기 힘들다는 점이 지적된다.

4

2. Classes of networked video service

2.1 download service

capture

display

encode

decode

transmission

- 고화질 콘텐츠 감상
- 손실 전혀 없고, 지연이 상관없음

5

2. Classes of networked video service

2.2 streaming : simultaneous transmission/replay of encoded video

- 서비스 제공자의 서버에 저장되어있는 비디오 콘텐츠를 사용자가 받으면서 동시에 실시간으로 볼 수 있는 서비스이다.
- 장점: 사용자로서는 보고싶을 때 바로 볼 수 있고, 서비스 제공자로서는 콘텐츠가 복사되는 것을 막을 수 있다.
- 단점: 송출 시스템이 복잡해지고, 통신망의 상태에 따라 서비스의 품질이 저하될 수도 있다는 단점이 있다.

6

2\. Classes of networked video service

2.2 streaming service : 예) VOD

capture display

encode decode

transmission

- 약간의 버퍼링 지연 허용(<10 sec)
- 패킷 손실을 감수해야함(<10%)

7

2\. Classes of networked video service

2.3 conversational: simultaneous encoding/transmission/decoding

- 화상전화, 화상회의, 감시시스템처럼 비디오가 캡쳐되면서 동시에 전송되는 서비스이다.
- 단점: 지연에 매우 민감한 서비스이며, 실시간으로 인코딩이 되어야 한다는 점에서 송출하는 시스템의 부담이 크다. 총 지연이 150ms 이상이 되면 사람들이 불편을 느끼게 된다고 한다. 그리고, 보통 인코더가 디코더 보다는 알고리즘이 훨씬 복잡하므로 실시간으로 인코딩하는 시스템은 가격이 비싸지게 된다. 가격대를 낮추려면 화질 저하를 감수해야 한다.

8

2. Classes of networked video service

2.3 conversational service

capture display

encode decode

transmission

- 화질보다 지연 줄여야(<150ms)
- 패킷 손실을 감수해야함(<10%)

9

End User QoS in UMTS

	conversational delay<<1sec	interactive delay 1sec	streaming delay<10sec	background delay>10sec
error tolerant	conversational voice and video	voice messaging	streaming audio and video	fax
error intolerant	telnet, interactive games	e-commerce, WWW browsing	FTP, still image, paging	e-mail arrival notice

10

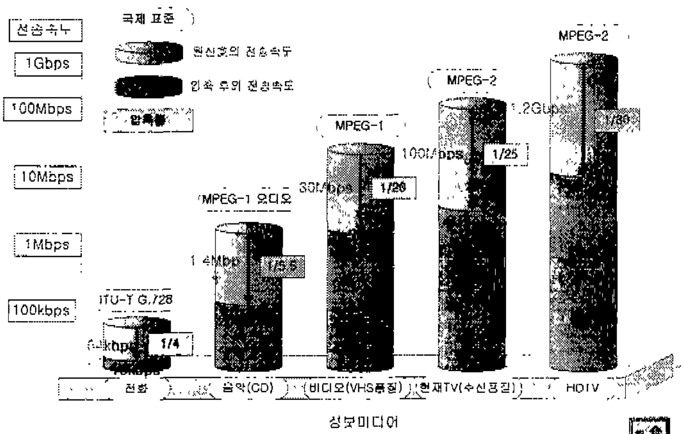
3. Video vs Audio

- Bandwidth hungry
- VBR for constant quality
- Temporal correlation

11

3\. Video vs Audio

3.1 Bandwidth hungry

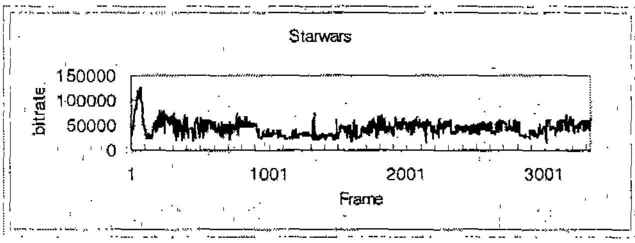


12

3. Video vs Audio

3.2 VBR or CBR

- Encoded audio and speech are CBR.
- VBR: 복잡한(단순한) 화면 비트율 크다(작다).
- VBR 가능하면 VBR로 (압축율 2배, DVD)



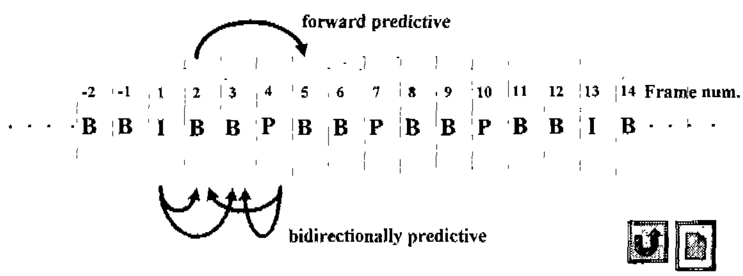
"Star Wars" encoded at VBR with constant quality

13

3. Video vs Audio

3.3 Temporal correlation

- Every audio packet is independent of each other.
- For higher compression ratio, ME/MC is used.
- 한 프레임 손상되면, 따라오는 여러 프레임 손상



14

4. Video codecs

All video codecs use DCT and ME/MC.

- MPEG-1 : for storage
- MPEG-2 : for storage and broadcasting
  - MP@ML : 3-6 Mbps, 720×576, 30 Hz
- MPEG-4 : multiple VOs and error resilience
- H.26x series : for video telephony
- Wavelet is effective on still images.

15

Various visual communication services

Application	Channel	Channel errors	Codec	Resol	Video bit u
Videophone	PSTN	Few bit errors and packet losses	H.263(+)	176X144	10-25 kb/s
Videophone	ISDN	No errors	H.261/263(+)	176X144	64-384 kb/s
Videophone	Packet	No bit errors, packet losses	H.261/263(+)	176X144	10-384 kb/s
Videophone	Wireless	High bit errors and packet losses	H.263(+)	176X144	10-300 kb/s
Videoconference	Packet	No bit errors, packet losses	H.263(+)	352X288	0.1-1 Mb/s t
Videoconference	ATM	Almost no errors	MPEG-2	720X480	1-6 Mb/s
DTV	Cable/satellite	Almost no errors	MPEG-2	720X480	4-12 Mb/s
HDTV	Terrestrial	A few bit errors	MPEG-2	1920X1152	18 Mb/s

* MPEG-4 applies whenever H.263 does.

16

5. Network QoS

- Goal of network :
  temporal and semantic transparency
- semantic QoS : loss
- temporal QoS : delay, jitter

❑ Loss와 delay는 서로 trade-off 관계이다.

17

Classes of networks

- Circuit switching : modem, IS-95AB
- Packet switching : Internet, LAN
  best effort(QoS not guaranteed)
- Cell(or label) switching : ATM, MPLS, full QoS negotiable

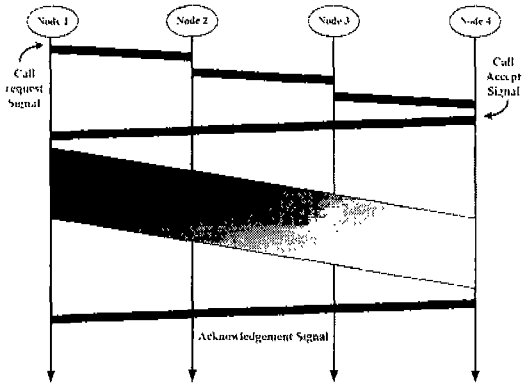
* UMTS : CS + PS + LS

18

Network QoS

5.1 Circuit Switching(회선교환)

- dedicated CBR channel, low delay/loss
- low bandwidth(64kbps in UMTS)

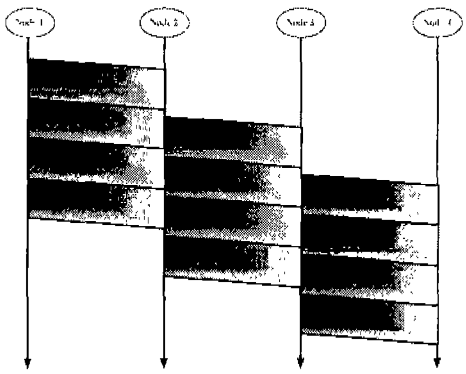


19

Network QoS

5.2 Packet Switching(패킷교환)

- shared best effort channel(VBR)
- time-varying QoS (congestion)

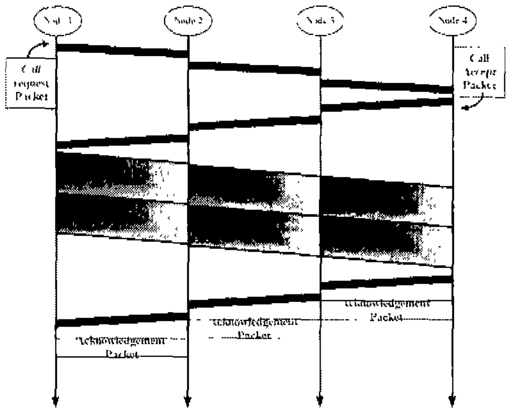


20

5. Network QoS

5.3 Cell(label) Switching

- Reserved QoS guaranteed (VBR)
- ATM, NGI(diffServ, RSVP, MPLS), UMTS



21

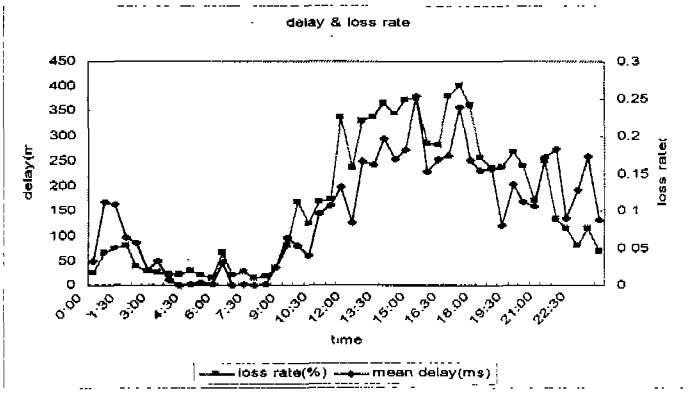
6. Quality of the current Internet

- best effort protocol
  - time varying QoS : loss, delay, jitter
  - Packet loss is bursty.
  - packet size dependent
- access network dependent
- routing delay because of long address and network congestion

22

6. Quality of the current Internet

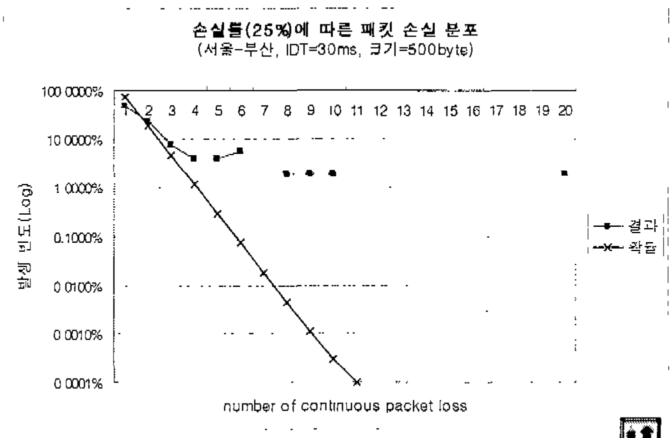
6.1 Time varying QoS (Suwon ⇔Pusan)



23

Quality of the current Internet

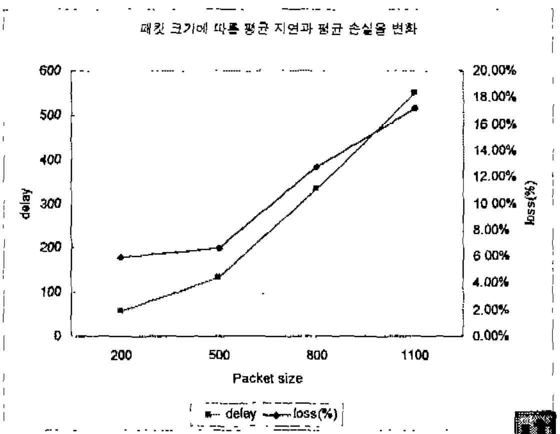
6.2 Packet loss is bursty.



24

Quality of the current Internet

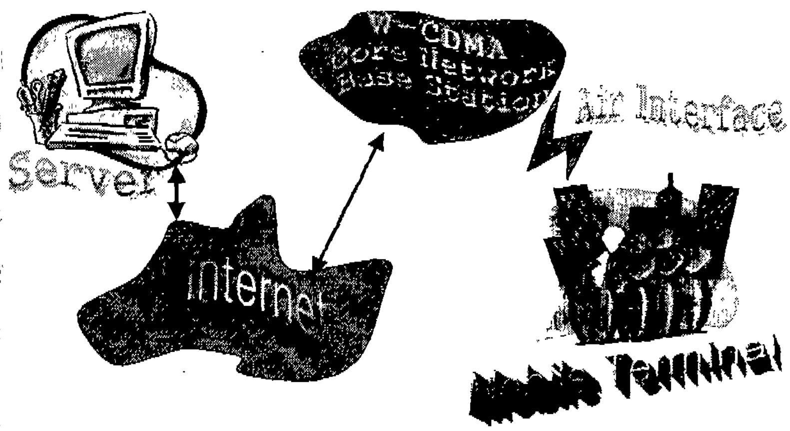
6.3 Packet size dependent



25

7. Quality of wireless Internet

- Internet + wireless network



26

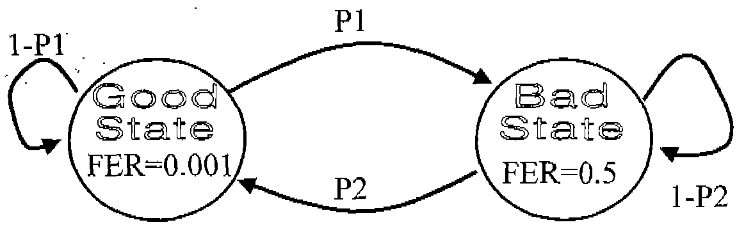
Video streaming over mobile Internet

- Key Issues
  - Available bandwidth
  - Error prone, time-varying QoS
  - Packet switching or Internet over circuit switching
  - Handover technology for video service
- 2.5th Generation (PCS)
- 3rd Generation (IMT-2000)
- 4th Generation (-2010)

27

7.1 Quality of wireless link

- Quality is dependent on ...
  - Distance between BS and MT
  - Time-varying : multi-path, speed
- Bursty, Gilbert model (NTT DoCoMo)



FER(frame erasure rate)

28

7.2 Value ranges for UMTS Bearer Service Attributes

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate (kbps)	< 2 048	< 2 048	< 2 048	< 2 048
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size (octets)	<=1 500 or 1 502	<=1 500 or 1 502	<=1 500 or 1 502	<=1 500 or 1 502
Delivery of erroneous SDUs	Yes/No/-	Yes/No/-	Yes/No/-	Yes/No/-
Residual BER	$5*10^{-2}$ 10^{-2}, $5*10^{-3}$ 10^{-3}, 10^{-4}, 10^{-6}	$5*10^{-2}$, 10^{-2} $5*10^{-3}$, 10^{-3}, 10^{-4} 10^{-5}, 10^{-6}	$4*10^{-3}$ 10^{-5}, $6*10^{-8}$	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$
SDU error ratio	10^{-2} $7*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}	10^{-1}, 10^{-2}, $7*10^{-3}$, 10^{-3}, 10^{-4}, 10^{-5}	10^{-3}, 10^{-4}, 10^{-6}	10^{-3}, 10^{-4} 10^{-6}
Transfer delay (ms)	100 – maximum	250 – maximum		
Guaranteed bit rate (kbps)	< 2 048	< 2 048		
Traffic handling priority			1,2,3	
Allocation/Retention priority	1 2,3	1 2,3	1 2,3	1,2,3

29

7.3 QoS of IS-95C, CDMA2000

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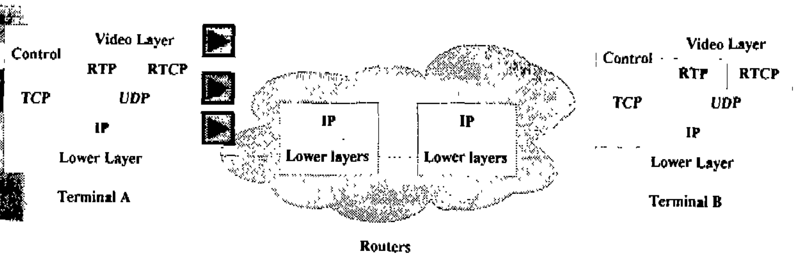
30

8. How to overcome?

- Each layer has its own roles.



IP Header	UDP Header	RTP Header	DATA

31

8. How to overcome?

8.1 Video layer

- error resilient coding : MPEG-4, H.263v2
- error concealment : It's up to developer.
- scalability(especially FGS) : MPEG-4

32

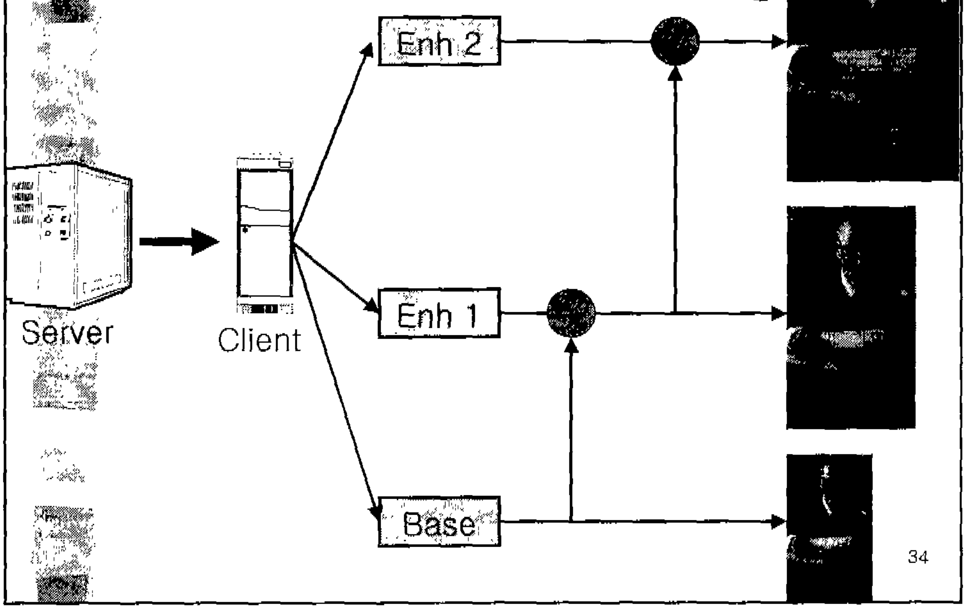
How to overcome?

8.1.1 Error resilience in MPEG-4

- RM(Resynchronization Maker)
- RVLC(Reversible Variable Length Code)
- DP(Data Partitioning)
- HEC(Header Extension Correction)
- AIR(Adaptive Intra Refresh)

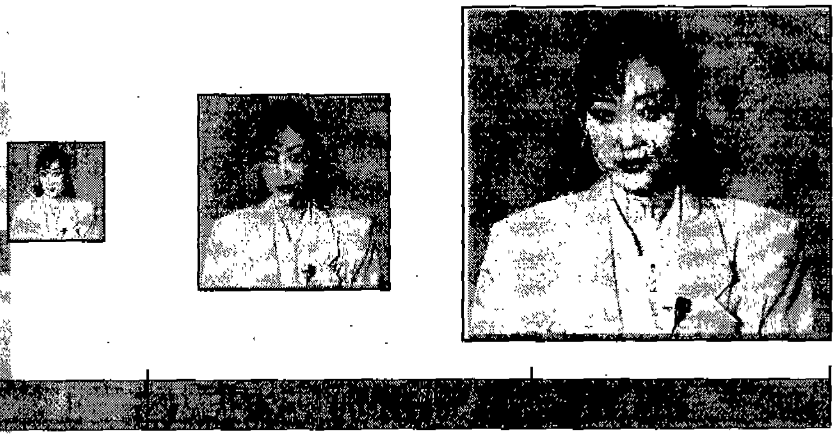
33

8.1.2 Scalable Encoding



34

Spatial scalability



14kbits 34kbits 47kbits

35

SNR scalability

5kbits 8kbits 30kbits

36

8.2 Transport layer

- Call setup : path, resources
- QoS control : RTP/RTCP
- QoS enhancement (not standardized, yet)
  - FEC with erasures
  - Interleaving
  - Dejittering buffer
  - Retransmission if allowed

37

RTP(realtime transport protocol)

0			8		16	31bits
V=2	P	X	CC	M	payload type	sequence number of primary

time stamp of primary encoding

synchronization source (SSRC) identifier

contributing source (CSRC) identifier

......

38

RTCP

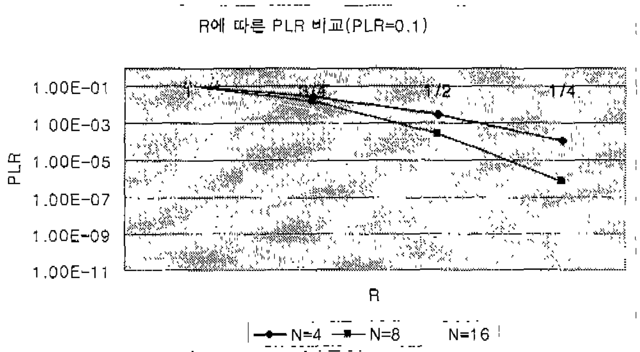
(realtime transmission control protocol)

- Measure network quality : loss, delay, jitter
  - SR (sender report)
  - RR (receive report)
  - SDES (source description items)
  - BYE
  - APP

39

FEC with erasure

- RS(10,6)을 사용하면, 데이터 패킷 6개, 패러티 패킷 4개를 전송하고, 10개중 아무거나 4개 잃어버려도 완전 복원 가능(RTP로 잃어버린 패킷위치 알 수 있음)
- IETF에서는 ULP(unequal loss priority) 표준화중임

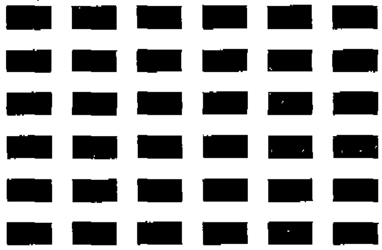


40

8.2.2 Interleaving

- to combat against bursty loss : loss rate vs. delay

Example) 1) No Interleaving : 2 out of each set of 6 packets
(1 2 3 4 P1 P2)(5 6 7 8 P3 P4)(9 10 11 12 P5 P6)(13 14 15 16 P7 P8)......
2) Interleaving : up to 12 consecutive packets

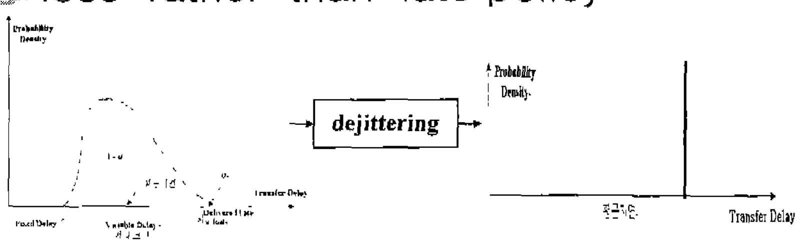


1 5 9 13 17 21 2 6 10 14 18 22 3 7 11 15 19 23 4 8 12 16 20 24 P1 P3 P5 P7 P9 P11 P2 P4 P6 P8 P10 P12

41

8.2.3 Dejittering buffer

- RTCP for measuring delay
- loss rate vs. delay
- loss-rather-than-late policy



Pdf befor dejittering ➔ dejittering ➔ pdf after dejittering

42

8.3 Network layer

- TOS(type of service) in IP header: not implemented in most routers (버스전용차선)
- QoS reservation : proprietary solutions of Intserv/RSVP or diffServ in Intranet or virtual LAN

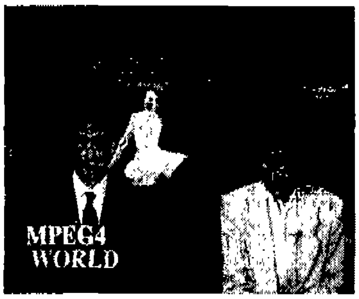
43

QoS Parameters of UMTS bearer and radio access bearer

- [Maximum bitrate(bps), Maximum SDU size(octet)] cf. [PCR, CDVT]
  - possible bitrates per subflow in CS case <= inter PDU transmission interval (IPTI)
- [Guaranteed bitrate, k*Maximum SDU size] cf. [SCR, BT+CDVT]
  - to capture burstness
  - minimum resource requirements => resource sharing
- delivery order(y/n)
- SDU format information(bits)
- SDU error ratio : loss+damaged
- residual bit error ratio
- delivery of erroneous SDUs(y/n/-) : '-' implies no error detection.
- transfer delay(ms) : 95% quantile
- traffic handling priority : per flow
- allocation/retention priority : per bearer which is not negotiated from the MT

44

Error Robust Video Service



ER(RM only) ER+FEC+Retransmission

45

9. Conclusion

- 서비스 종류에 따라 요구되는 QoS가 다르다.
- 교환방식 : Circuit => Packet => Label
- 교환방식에 따라 QoS 특성이 다르다.
- 프로토콜 계층(layer)마다 가능한 QoS 향상 방법을 모두 사용하여야 한다.
- market pull
  - 비디오에 익숙한 세대, '인디'의 시대
- technology push
  - broader bandwidth, QoS guaranteed network

46

약어표

- ATM : asynchronous transfer mode, B-ISDN
- BER : bit error rate
- BS : base station
- CBR : constant bit rate
- DCT : discrete cosine transform
- Enh : enhancement layer
- FER : frame erasure(loss) ratio
- FGS : fine granular scalability
- IETF : Internet engineering task force
- ME/MC : motion estimation and motion compensation
- MPLS : multiple protocol label switching
- MT : mobile terminal
- QoS : quality of service
- RS : Reed-Solomon code
- RSVP : resource reservation protocol
- RTP/RTCP : realtime transmission protocol / realtime transmission control protocol
- SDU : service data unit
- SNR : signal to noise ratio
- UMTS : universal mobile telecommunication system : - 3GPP, IMT-2000
- VBR : variable bit rate
- VOD : video on demand

47

Quality of Service

end-to-end(TE-to-TE) quality perceived by the customer(ITU-T M.xxx)

the collective effect of service performances such as;

- service operability performance;
- service accessibility performance;
- service retention performance;
- service integrity performance; and
- other factors specific to each service.

48