IKT Convergence Pre-Project: Digital Video Broadcast

Norwegian Computing Center
APPLIED RESEARCH AND DEVELOPMENT
Norwegian Computing Center / Applied Research and Development

NOTAT/NOTE

Norges forskningsråd

IMEDIA/09/99

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Sammendrag/Abstract:

Telecom-2005 — IT-næringens teleforskningsforum prosjekt "IKT konvergensteknologier" — consists of five pre-project packages covered by different groups of partners. The Siemens, Telecast, NR and SINTEF group has been working in the area of Digital Video Broadcast (DVB).

Within the area of Digital Broadcasting, there is a need for solutions based upon standard network technologies such as IP, ATM and SDH. This pre-project, completed in 1999, has identified the standard communication technologies which can be used in distribution and contribution of DVB signals. While ATM is today the state-of-the-art technology for DVB distribution and contribution networks, the pre-project results have shown that IP has the potential to become the technology-of-the-future for both contribution and distribution of video. The pre-project has further identified the shortcomings of IP which need to be overcome before IP can attain widespread use for DVB.

The the explicit goals of the pre-project have been: (1) to employ competence and resources within Norwegian industry, development and research environments, in order to (2) identify and investigate critical areas of technology with regard to the use of standard data transmission technologies for digital TV transmission; and, (3) to describe goals, possible results and resource requirements for a main project within IKT Convergence.

The written results documented herein are presented as technical summaries and presentations. These materials address the following major areas:

- QoS Mechanisms and Network Usage
- DVB-T over ATM and IP
- DTV production and transmission
- QoS in IP-based Networks
- Users and interactive TV.

Emneord/Keywords: DVB, ATM, IP, QoS, interactive television

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IKT Convergence Pre-Project: Digital Video Broadcast

Final Report

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Table of Contents

IKT Convergence Pre-Project:
Digital Video Broadcast

1. Objectives

Within the area of Digital Broadcasting, there is a need for solutions based upon standard network technologies such as IP, ATM and SDH. This pre-project, completed in 1999, has identified the standard communication technologies which can be used in distribution and contribution of DVB signals. While ATM is today the state-of-the-art technology for DVB distribution and contribution networks, the pre-project results have shown that IP has the potential to become the technology-of-the-future for both contribution and distribution of video. The pre-project has further identified the shortcomings of IP which need to be overcome before IP can attain widespread use for DVB.

1.1 Siemens Norway

Siemens Norway has been given the responsibility to establish a Center of Competence (CoC) within the area of DVB. CoCs within Siemens function as technical advisors for other Local Siemens companies which need expertise when preparing technical solutions and tenders to local customers. The objective behind Siemens Norway within this pre-project is to:

- establish close relations between the CoC in Siemens and expertise in universities or research institutions on the area of DVB in Norway,
- use the CoC competence together with the national expertise and competence to recommend, develop and deliver DVB customers solutions world wide.
- prepare for use of new technology as IP in DVB distributions.

1.2 TeleCast Norway

TeleCast Norway supplies the broadcasters and the telecommunication companies with analogue and digital routers and switches. From our normal study of market needs we see a possible change in ways of distributing and contributing that will affect our product range and knowledge in not-so-distant future.

The objectives for TeleCast Norway are therefore:

- to increase our competence in the field of communicating via networks, and
- to find the impact this will have on our product range strategies.

1.3 SINTEF

Sintef Telecom and Informatics has been involved in digital TV broadcast technology and MPEG/DVB for several years, as well as technologies for high capacity networks and broadband ISDN (ATM). Being a contract research partner for (mainly) Norwegian companies, Sintef aims at being up front with respect to knowledge and development of new technologies.

In the area of technology convergence, Sintef acknowledges the integration of information, communication and broadcast technologies, and sees this Convergence Project as a means for both increasing its own competence and for contributing to relevant technology development.

Areas of special interest are

- Modulation, compression, coding and synchronisation for digital video transmission and broadcast.
- Utilisation of network capacity in a heterogeneous traffic and technology environment, as well as the use of QoS mechanisms for serving different traffic classes.

1.4 Norsk Regnesentral

For a number of years, Norsk Regnesentral's (NR) departments for information technology have been carrying out applied research in the areas of object-orientation, data communication, organizational development, distributed systems, interactive media and security. Since the inception of WWW technology, NR has participated in an ever-increasing percentage of projects concerning state-of-the-art use of media streaming, security and distributed system technologies. Through the IKT Convergence project, NR is looking to apply and further its more specialized competence in the areas of media and transmission formats, quality of service, design of interactive applications and services, digital television, multimedia production, systems design, and user interface and usability issues.

2. Pre-Project Goals

Given the set of objectives listed above, the explicit goals of the pre-project have been:

- to employ competence and resources within Norwegian industry, development and research environments, in order to:
- identify and investigate critical areas of technology with regard to the use of standard data transmission technologies for digital TV transmission; and,
- to describe goals, possible results and resource requirements for a main project within IKT Convergence.

3. Process / Approach

The pre-project has been carried out during Q3-Q4 1999. The resource frame for this pre-project is presented in the table below.

Table 1

	Timer	100 % NOK/T	1/3 dek. NOK/T	Total	NFR Bidrag	Bedrifter Egenandel
Institutt	200	750		150 000	150 000	
Siemens, Telenett	150	750	250	112 500	37 500	75 000
TeleCast Group	100	750	250	75 000	25 000	50 000
Partner NN	100	750	250	75 000	25 000	50 000
Institutter	200			150 000	150 000	
Bedrifter	350			262 500	87 500	
Reiser					50 000	
Prosjekt Totalkostnad				412 500		
Finansiering					287 500	175 000

Given this temporal and economic resource frame, the scope of the pre-project was limited to DVB-T and digital TV contribution and distribution over terrestrial networks. With this scope, the project was divided into four subprojects, each having a specific area of focus.

Each subproject was led by one of the four different project partners. Through group discussion, fundamental technical / functional issues and areas were identified and prioritized within each of the subprojects (see below and section 6). Thereafter, these issues were investigated and reported upon within the project; this project-internal reporting was carried out through the writing, distribution and presentation of technical summaries and other materials.

3.1 SINTEF — *QoS Mechanisms and Network Usage*

Digital video streaming can be transferred in an ATM or IP based network using either dedicated or shared connections. In case of shared connections with mixed traffic one needs to look at QoS mechanisms to ensure proper treatment of the video signal with respect to delay and data loss. Within this subproject, the problem is briefly described, and possible solutions indicated, for offering QoS in mixed traffic networks. Some thoughts on the future regarding digital production facilities built upon ATM or IP devices are also provided.

3.2 Siemens — *DVB-T over ATM and IP*

As supplier of network solutions Siemens focused in the project on network issues and transmission of the DVB signals between DV producers and distribution of DVB to transmitter sites. Issues in Siemens contributions are therefore: synchronization, quality of service error correction, routing in IPv6 and network management.

3.3 TeleCast *DTV production and transmission*

TeleCast, as a supplier of routers and control software to DTV producers and distributors, has focused on investigating the views of these producers and distributors regarding future communication of semi-products and end products, in-house and external. Issues for TeleCast have therefore been to interview producers and distributors about their views on ATM and IP as communication network.

3.4 Norsk Regnesentral *QoS in IP-based Networks; Users and interactive TV*

NR has competence and project expericence in the area of seamless networks and interactive television. Thus, NR's contributions in this pre-project have focused in the area of QoS in IP-based networks, as well as with application- and end-user level issues arising as a result of the emergence of interactive television.

4. Results

Of essential value, the project has resulted in a common base of competence, understanding and terminology within the project team. The written results from the project include the technical summaries and presentation which are listed below and included in the following Appendices:

- Optimalisering av Sambandsutnyttelse ved Digital Videotransmisjon over ATM og IP
- Distributed Video Production
- HDTV in Internet2
- Synchronisation of SFN in IP Networks
- Multi Protocol Label Switching
- Advantages for IPv6 in relation to DVB and DVD
- Network Management
- Influences of Radio Relay parameters upon ATM QoS and video quality
- Digital audio in professional applications
- Krav til Forsinkelse og Bildekvalitet
- Tjenestekvalitet ved distribusjon av digital video over IP/ATM nettverk
- Interactive Television

These Appendices are included as part of this document, to act as a record for project work thus far, as well as a reference for work ahead.

Lastly, the project has developed a preliminary draft of a project specification — a specification for a follow-up project within IKT Convergence.

5. Future Work

The work ahead concerns the creation of a new, main project in the area of DVB and network communication technology. The basic outline of the proposal has the unifying objective of investigating and determinating solutions for use of new technology in DVB and developing the technological basis for establishing DVB as the new TV distribution technology.

The primary themes of the proposal, spanning a period of 2 to 3 years, are:

- To solve the outstanding issues related to use of ATM in distribution and contribution, using a test network as an experimental DVB platform
- To design, develop and evaluate new interactive services delivered over DVB, and
- To monitor technological evolution, especially the balance between ATM and IP within network infrastructures.

Possible activities related to each of these themes are presented below.

5.1 Trial network test environment

A DVB test and development environment in Oslo for contribution of DV and distribution of DVB and establish ATM trial network with SFN transmitter test sites.

Possible activities:

- Specify QoS requirements for distribution and contribution
- Technology evaluation for heterogeneous networks.
- Comparing transmission cost for TV distribution/contribution.

5.2 ATM in contribution and distribution

The advantage of ATM is the QoS definitions and the ability to deliver cells with a specified delay, cell loss and variable cell delay. For real time transmission and interactive TV production with a dialogue between persons in separate studios, ensuring bounded delay is important.

Proposed activities:

- ATM network for Contribution and Distribution and use of AAL1 or AAL5
- Compare QoS functions in ATM and IP
- FEC correction mechanism in N/A for ATM for transport
- Resynch and signal restoration in SFN networks based on ATM

5.3 DVB production and distribution of interactive TV services

This will include activities in the project covering both the transport of signals from TV-production into distribution out to subscribers and the communication over the return channel that can be based on ordinary ISDN or cable technology.

The goal can then be to define a set of possible services that can be delivered over DVB and find a solution for IP based DVB network with interactive communication based on Set Top box or TV-set.

Proposed activities:

- User role and New Services in DVB Distribution.
- Prototyping and definition of Services for DVB network
- Return channel network solutions for Interactive Services
- Test of interactive Services based on Set-Top boxes and return channels
- Trial period with customer

5.4 IP in contribution and distribution

In contribution where real time analog/digital transmission is used, we will se a much more differentiated use of standard data communication technology. For all transport of file based video, both faster and slower than real time, IP can be used.

IP networks with Packet Over SONET (POS) or Packet over WDM can be installed with sufficient capacity to transport 270 Mbit/s raw video signals and can be used down to 2 Mbit/s for transport of compressed video programs. One project activity can therefore be to specify and test solutions for use of IP in contribution of TV programs.

Proposed activities

- Feasibility study of DVB IP based network addressing issues such as bit error rate in RL
- IP over ATM, SDH or WDM for transport in DVB networks
- Test and evaluation of QoS functions in IPv6 networks
- Multicast in IPv6 networks
- Restoration of signal transport in SFN network with IP
- IP in distribution of DVB
- Trial Network with DVB over IP.

APPENDICES

6. Itf Konvergens prosjekt - DVB Siemens/TeleCast
Prosjektplan forprosjekt

Prosjekt organisering

Prosjektet er inndelt i fire delprosjekter fordelt på de fire partnerene:

SINTEF Standarder og Internet 2
Siemens DVB-T over ATM og IP
TeleCast DTV produksjon og overføring
Norsk Regnesentral Brukere og interaktiv DVB

Delprosjektene er inndelt i deloppgaver med beskrivelse av resultatet fra hver deloppgave ("deliverable") og med en ansvarlig for oppgaven.

Prosjekt rapporten være et utkast til et nytt prosjekt og resultatene fra hver del-oppgave brukes som vedlegg i utkastet til et hovedprosjekt

<u>Det er viktig nå å få en oversikt over hva som kan gjennomføres innen forprosjektet rammen (budsjettet).</u>

Tabellforklaringer

"deliverable" skal beskrive så konkret som mulig hva som blir bidraget til sluttrapporten for aktiviteten.

Prioritet skal vise rekkefølgen for aktiviteten (prioritet) og når den er planlagt fullført.

Utføres av kolonnen viser hvem som utfører oppgaven og eventuelle bidrag fra andre og antall timer planlagt for oppgaven (brukes for rapportering/eventuelt grunnlag søknad om utvidelse av timerammen)

Eventuell oppgaver som skyves til hovedprosjekt kommenteres her.

SINTEF	Beskrivelse	"Deliverables"	Prioritet Fullført	Utføres av
Standarder				
Sintef-A: Gjennomgang dokumenter fra ACTS-DVP og ETSI.	Trekke ut den informasjon som er relevant for vårt prosjekt og presentere dette som et "technical summary". Dette innebærer en to-trinns gjennomgang, først for å skaffe seg oversikt i dokumentene som er utgitt, dernest en mer detaljert pløying i et fåtall aktuelle dokumenter. Vi er ute etter både dagens standarder og fremtidig utvikling, hvor sammenligninger mellom ATM og IP er av spesiell interesse (hvis det fins).	Presentert som et "technical summary".	Innholdsovers: 29.10 Draftversjon: 15.11 Sluttversjon: 29.11	SINTEF Omfang 30 timer.
Sintef-B: Beskrive mulige tilleggsdata og hvordan dette kan overføres over IP eller ATM forbindelser samtidig med MPEG/DVB.	Her vil det være nødvendig med en brukerdialog for å kunne beskrive behovet for overføring av data. Eksempler er internettrafikk, filoverføring og elektronisk post, som ikke er svært sensitive på forsinkelser og sperr/pakketap. Behovet må så sammenholdes med DVB transmisjonsbehov og tilbudt linjekapasitet.	Del-1 Hovedpunkter i en"brukerundersøkelse", samt innholdsfortegnelse Del-2 Technical summary med forslag til demonstrator/simulering av realistiske trafikk sammensetninger.	Del-1: 29.10 Del-2 Draft versjon: 15.11 Sluttversjon: 29.11	SINTEF Omfang: 30 timer.
DV over Internet 2				
Status DTV over Internet2 Undersøke arbeidet som gjøres i Internett2.	Det er demonstrert at IP kan overføre 270 Mbit/s over Internet 2. Hva er status for denne aktiviteten i USA	Oppsummering av Status i tillegg til presentasjon på prosjektmøte	Ferdig	SINTEF

Siemens	**Beskrivelse**	**"Deliverables"**	**Prioritet Fullført**	**Utføres av**
DVB-T over ATM og IP				
IP og Synkronisering av Single Frequency Network,	Distribusjons nett for SFN sendere har sterke krav til oppetid for sendere og at utfall på mer enn 100ms ikke kan aksepteres av DTV distributørene. Dette betyr at det settes store krav til om ruting og krav til resynkronisering av nettet, kodere og nett adaptere ved utfall av en transmisjons strekning. En fordel ved IP er at nettlaget er "connection less" i motseting til ATM som er "connection oriented" dvs at ATM må sette opp forbindelsen på nytt ved brudd, mens IP nettet automatisk velger en alternativ vei for IP pakken.	Beskrivelse av problemet med krav til IP nett for SFN	2 Nov. 15 first draft Nov. 29 final draft	Siemens i samarbeide med av SINTEF
Feilbit rate i radio linker behov for Forward Error Correction FEC ved IP overføring Eventuelle andre metoder for feiloppretting med bruk av TCP/IP	Forward Error Correction (FEC) benyttes i N/A for å kompensere for celletap. Dvs at N/A kan rette opp en gitt mengdefeil i en MPEG-2 pakke forårsaket av celletap i ATM cellestrømmen. ATM har korte celler (48 byte payload), mens IP benytter pakker med variabel lengde opptil eks 4K byte. Tap av en IP pakke er derfor vanskeligere å rette ved hjelp av FEC	Oppsummering og vurdering av løsninger for feilkontroll og FEC i IP nett	3	Flyttes til hovedprosjekt

Ruting i IP nett, og bruk av MPLS løsninger for å redusere pakke forsinkelse i hver node. Status for implementering av MPLS, tilgjengelighet og driftserfaringer	Innvendingen mot IP brukt i "real-time" overføring som f.eks VoIP og Multimedia, er at adresse oppslagene i hver node tar tid og forsinker IP pakken i hver node den passerer. Multi Protocol Label Switching er en metode for å realisere connection orientert switchnig i IP noden og derved redusere forsinkelsen i hver node ved at pakken videresendes ifølge en forbindelses tabell uten adresseoppslag. MPLS kan derfor fjerne en vesentlig del av tidforsinkelsen i IP nett og ved bruk av dedikerte IP nett kan de være mulig å ha overføring i virkelig tid	Oppsummering av status for MPLS	3 Nov. 15	Nils
Har IPv6 noen fordeler i distribusjon slik som prosjektet definerer distribusjon. F.eksempel gjennom multicast adresser for overføring fra en mange og bruks av høyhastighets LAN for lokal overføring.	IPv6 er tenkt for blant annet å distribuere video og lyd over internet. IPv6 er sagt å ha bedre adresse egenskaper ved overføring fra en til flere/mange. Ved overføring mellom produksjonssteder kan det være nyttig å benytte IP over 100 Mbit Ethernet LAN / Gigabit Switcher for lokal overføring mellom studioer eller for overføring til andre studio/produsenter. Hovedbegrunnelsen for å bruke IP er at IP har lavere kostnad enn ATM og kan benyttes av mindre produksjonssteder.	Oppsummering av egenskaper til IPv6 i relasjon til bruk i distribusjon	1 Nov. 15	Nils med ass. fra NR
Defining and requirements to combined network Management solutions	Is it possible to have one management solutions or must there be two different NMS by definition, one for the net, the other for the transmission/Video.	Krav til driftsløsninger og forslag til felles NMS (TMN) system	2 Nov. 15	Frank
Influences of Radio Relay parameters on the QoS ATM, exspecially under faulty conditions and the limit of NA to recover good MPEG TS	Radio overføring kjennetegnes ved gradvis økende feilbitrate på grunn av atmosfæriske forhold. N/A har forskjellige egenskaper for å kompensere for feilbit i radio linje overføring. Variable Cell Delay tas med som tema i arbeidet.	Forslag til mulige tester og forsøk	1 Nov.15	Frank

Undersøke typiske/mulige kvalitetskrav i ATM forbindelser. USA foretrekker å bruke AAL5, og i Europa brukes CBR og AAL1?	Finne ut hva brukerne forventer, og hvilke krav som stilles. Er det forskjeller mellom USA og Europa mhp. DVB transmisjons nettet eller andre krav som avviker mellom USA og Europa	Kort oppsummering av problemstillingene rundt AAL1 eller AAL5	3 Nov. 29	Flyttes til hovedprosjekt

TeleCast	**Beskrivelse**	**"Deliverables"**	**Prioritet Fullført**	**Utføres av**
DTV produksjon og overføring				
Hva er Kringkastings- og Telecom selskapenes syn på ATM og IP? Det er viktig å danne seg et bilde av hva selskapene vil. Hvilke inntrykk fikk en på IBC ?	Kringkastings- og Telecom selskapene er våre kunder og det er nødvendig for vårt strategi arbeide å ha klarhet i hva kundene tenker på. Vi må da skaffe oss kunnskap for å delta i diskusjonen om løsninger de trenger.	Rapport	Nov. 15 first draft Nov. 29 final draft	TeleCast med ass.fra Schmull
Muligheter for bruk av IP i produksjons overføringer. Hva skiller ATM og IP mhp. fordeler og ulemper i produksjons overføring/distribusjon	Produsenter og DTV distributører vil ha best mulig løsning til billigst mulig pris. IP er et kostnads gunstig alternativ i forhold til ATM	Oppsummering av dialog med NRK/TV2 eller liknende	Nov. 15 first draft Nov. 29 final draft	TeleCast
Undersøke krav til forsinkelse og bildekvalitet ved distribusjon mellom DV produsenter. Hvilke krav sette de forskjellige produksjonsmetoder	Tidligere diskusjoner med kunder har vist at de er opptatt av forsinkelse og synkronisering. Dette trenger kvantifisering	Rapport	Nov. 15 first draft Nov. 29 final draft	TeleCast
Rapportere status for standardisering av lyd overføring i ATM nett for profesjonell bruk	Det foreligger internasjonalt et initiativ for å standardisere lydoverføring over ATM. TeleCast vil følgeprosessen og rapportere til prosjketet	Oppsummering av Status	Nov. 29 final draft	TeleCast

Norsk Regnesentral - NR	**Beskrivelse**	**"Deliverables"**	**Prioritet Fullført**	**Utføres av**
Brukere og interaktiv DVB				
NR-a: Determine status of QoS activities, approaches, etc., for both ATM and IP. Compare QoS in ATM with possibilities in IPv4 / IPv6. Focus upon technical contexts in which infrastucture is dedicated rather than public.	DV production,distribution and/or broadcast may employ data links for DV transmission. Depending upon network load, there may be a need to provide guarantees (i.e., upper bounds) upon transmission delay, loss, etc. Alternatively, there may be a desire to prioritize amongst the different kinds of network traffic.	Technical summary	1 Oct. 29 Table of Contents Nov. 15 first draft Nov. 29 finaldraft	NR, Lars (ca. 40 hrs.)
NR-b: For interactive TV, consider the end-users (i.e., the viewer's) role, possible areas of use / use contexts and possible kinds of desired services.	Consideration of viewers' possible role, use contexts and desired services may generate business ideas which could ensure greater market margins.	Annotated presentation.	2 Nov. 15 first draft Nov. 29 final draft	NR, Peter (ca. 30 hrs.)
NR-c: Consider how aspects of viewer role/use might impact upon broadcast, distribution and production and/or the nature of DTV signals.	Consideration of viewers' possible role, etc. may generate requirements for broadcast and pre-broadcast processes / signalling.	Technical summary	3	NR ++ (resources out of scope; a task for 'hovedprosjektet'))
NR-d: (Related to point above:) For interactive TV, investigate how the return channel might be implemented and used. Consider how it might be possible to lead the *information* within the return channel as far as possible back into the production/distribution/broadcast process chain.	Business possibilities and technical requirements. See two points above.	Technical Report	3	NR ++ (resources out of scope; a task for 'hovedprosjektet'))

NR-e: Review 'Konvergens' document (NOU 1999:26)	This document provides: (1) a *very* wide overview of convergence-relevant technologies, as well as an overview of national and international activities / actors; and, (2) deliberations upon the existing regulatory frameworks for telecom, broadcasting and IT, and suggestions / recommendations as to how these frameworks need to be modified in light of the convergence of these technical areas.	(1) Annotated presentation; (2) Annotated presentation or technical summary;	3 Nov. 29th, Possibly "bulleted list" derived from work /study NR-e Part 1.	NR, Peter (Part 1: ca. 30 hrs.; Part 2: ca. 50 hrs.) Some of Part 1 may be achieved in the pre-project. Part 2 is a task for the 'hovedprosjekt'.

7. Optimalisering av Sambandsutnyttelse ved Digital Videotransmisjon over ATM og IP

7.1 Description of the topic

I det tilfellet at digitale videosignaler overføres i et ATM- eller IP-nett, kan det være hensiktsmessig å se på flere trafikktyper på samme forbindelse. Overføring av bredbåndsdata over leide linjer er kostbart. Derfor vil det være av stor interesse å kunne utnytte forbindelser for digitale videosignaler best mulig, når de først er opprettet, noe som i praksis vil bety å inkludere overføring av internettrafikk som filoverøring, e-post og web-aksess i samme forbindelse. Et eksempel på dette er vist i Figure 1. ATM-nett og IP-nett vil ha like prinsipper, men antall trafikkategorier vil kunne være forskjellige.

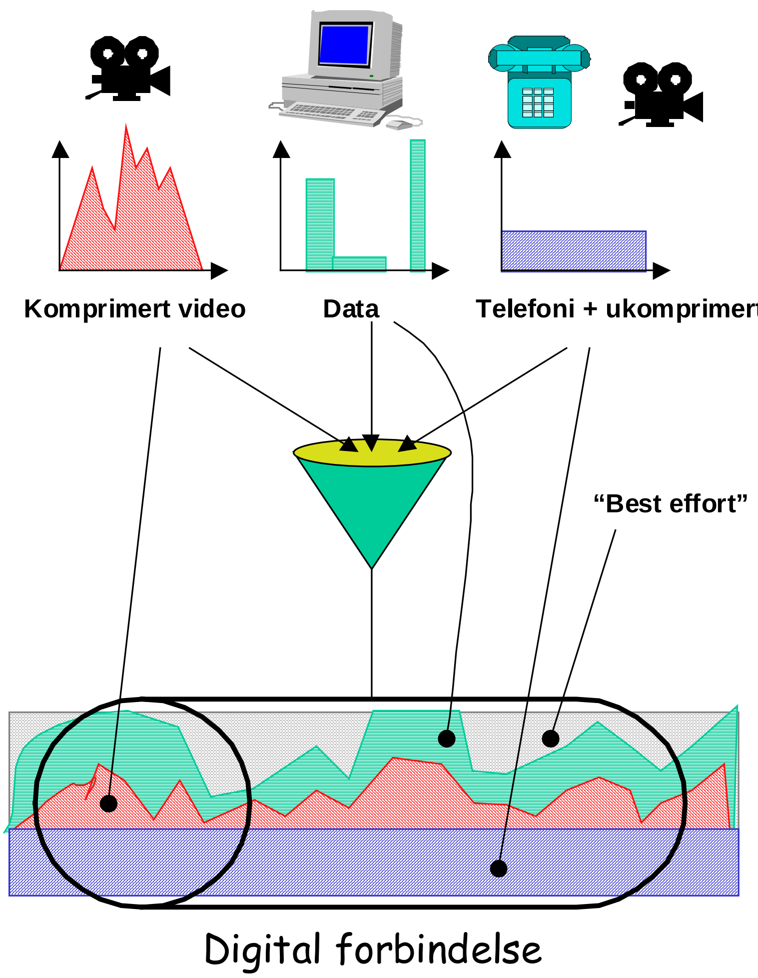

Figure 1. Illustrasjon av heterogen trafikk over et digitalt nett.

Man kan se for seg to hovedalternativer for utnyttelse av kapasitet i leide linjer, delt eller integrert løsning for digital video og internettrafikk. I tilfelle delt løsning vil man overføre kun videosignalet på dedikerte linjer, og internettrafikk vil gå gjennom en ISP eller over LAN/WAN. Ved integrert løsning ser man for seg å utnytte mekanismer for tjenestekvalitet slik at videosignalet får reservert tilstrekkelige ressurser, samtidig som det skal være tilleggskapasitet for ordinær internettrafikk i det samme nettet. I det ekstreme tilfelle kan man også se på overføring av *flere* videokanaler, både med og uten komprimering, i samme nett som internettrafikk.

Digital video kan transporteres mellom to lokasjoner som filoverføring eller som streaming. Det er i det videre antatt at streaming-format benyttes, filoverføring regnes som triviell, da den ikke stiller samme strenge krav til tjenestekvalitet. (Se kapittel 17 om tjenestekvalitet)

7.2 Major issues and challenges

Videosignalet danner en bitstrøm med en av følgende to egenskaper:

- 2 - 50 Mbps MPEG-2 kodet/komprimert signal (varierende bitrate over tid)
- 270 Mbps ukodet signal (konstant bitrate)

Av hensyn til forsinkelser som introduseres i kodeprosessen, samt krav til at det skal kunne klippes i en videostrøm etter hvert enkelt bilde er det fra NRK's side ikke anbefalt å gå lavere enn 50 Mbps for komprimert video i studiosammenheng. Ut til sluttbrukere, derimot, er det vanlig å sende et signal som er komprimert ned til ca. 6 Mbps.

Nettverket som skal transportere videosignalet er ett av følgende:

- ATM-nett med tjenestekategorier som CBR, rt-VBR, nrt-VBR, ABR og UBR.
- IP-nett utvidet med mekanismer som IntServ, RSVP, MPLS, DiffServ, RTP, eller flow label (IPv6).

Ved overføring av video i et digitalt nett må det stilles samme krav til nettet som ved overføring over dedikerte linjer. Det betyr igjen at reservasjonsmekanismer må fungere på en slik måte at den kapasiteten som er satt av til video oppleves som dedikert.

7.2.1 ATM

I ATM-nett er mekanismer for tjenestekvalitet veldefinerte og etablerte. For hver forbindelse som opprettes inngås det en trafikkontrakt mellom bruker og nett, og den trafikken som sendes over nettet blir hele tiden målt opp mot kontrakten. En anbefalt inndeling av ulike typer trafikk i ATM tjenestekategorier er gjort gjennom anbefalinger fra ATM Forum, og er vist i Table 2. For en oversikt over kategorier og parametre i ATM, se [1].

For streaming av digital video vil det trenges CBR for ukomprimert (konstant bit-rate) signal, og rt-VBR for komprimert (variabel bit-rate) signal. Begge videoformater trenger absolutte reservasjoner av nødvendig båndbredde.

Det er i Table 3 ikke sagt noe om generell internettrafikk. For transport av http- og ftp-type data er UBR egnet. Denne kategorien tilsvarer internettets "best-effort" tankegang.

Noen utfordringer i forbindelse med bruk av ATM for heterogen trafikk:

- Undersøke hvordan et nett reagerer ved forskjellige typer overbelastning.
- En sammenligning av SVC (svitsjet forbindelse) og PVC (permanent forbindelse) for overføring. Dette betyr i praksis om en forbindelse må forhåndsbestilles eller om den kan kobles opp automatisk ved behov.
- Aksesshastigheter fra kunde inn til transportnettet er i dag begrenset til 2, 34 eller 140/155 Mbps, på grunn av eksisterende tilkoblingsstandarder. Hvordan kan dette utnyttes best mulig? (Kan også sees i sammenheng med forrige punkt).
- Se på ADSL/VDSL teknikker for aksessnettet. Dette er teknikker for utnyttelse av kobberkabel (abonnentlinjer for telefon) for høyhastighets digital overføring.

	CBR	rt-VBR	nrt-VBR	ABR	UBR
Critical data	● ●	●	● ● ●	●	○
LAN interconnect	●	●	● ●	● ● ●	● ●
WAN transport	●	●	● ●	● ● ●	● ●
Circuit emulation	● ● ●	● ●	○	○	○
Telephony, Video-conferencing	● ● ●	○ ○	○ ○	○	○
Compressed audio	●	● ● ●	● ●	● ●	●
Video distribution	● ● ●	● ● ●	●	○	○
Interactive multimedia	● ● ●	● ● ●	● ●	● ●	●
● ● ● Optimum; ● ● Good; ● Fair; ○ Not suitable; ○ ○ Under review					

Table 2. Inndeling av forskjellige trafikktyper i tjenestekategorier for ATM [1].

7.2.2 IP

Et problem med bruk av mekanismer for tjenestekvalitet i IP-nett er at de ikke er endelig fastlagt, og ikke implementert i særlig grad i kommersielle nettverk. Hovedmekanismer for tjenestekvalitet og håndtering av tjenesteklasser er per i dag IntServ og DiffServ, se kapittel 17 om tjenestekvalitet for nærmere beskrivelse. I tillegg fins MPLS, som er en annen og uavhengig mekanisme som kan utnyttes for differensiering av datastrømmer.

IETF arbeider med standardisering på området, noe som betyr at det kan komme endringer i dagens anbefalinger og praksis. I Table 3 er angitt en måte å inndele trafikken i et IP-nett, med bruk av mekanismer fra IntServ/RSVP eller DiffServ. For nærmere forklaringer, se [2].

Trafikktype	IntServ/RSVP QoS	DiffServ QoS
Sanntidsdata (video, audio,..)	Guaranteed QoS	Expedited Forwarding (EF)
LAN sammenkoblinger	Controlled Load	Assured Forwarding (AF)
Internettrafikk	Best Effort	Best Effort

Table 3. En mulig fordeling av trafikktyper i et IP-nett.

Utfordringene med bruk av tjenestekvalitet i et heterogent IP-nett vil være mange av de samme som i ATM. Siden dette er et uferdig område fra IETF vil det være viktigere med uttesting av mekanismene i praksis for IP enn for ATM, særlig med tanke på ende-til-ende tjenestekvalitet.

I det tilfellet at en forbindelse strekker seg over segmenter fra flere nett- eller tjenesteleverandører er det verdt å merke seg at tolkningen av innholdet i de ulike tjenesteklassene kun gjelder innenfor *eget* leveranseområde.

7.2.3 In-house produksjon

I produksjonsfasen innenfor egne lokaler er det andre faktorer som spiller inn for kvalitet. Når det gjelder LAN er ikke båndbredder et problem slik som ved transport over større geografiske avstander.

Det finnes i dag mange formater for videosignaler, både digitale og analoge, med tilhørende matriser for svitsjing av signaler. På sikt ser man for seg at et produksjonsnett vil bli heldigitalt, flere produsenter har allerede kommet med ATM-utstyr. Når videosignalet foreligger på ATM format betyr det at en mer eller mindre standard ATM-svitsj vil kunne gjøre jobben som dagens matriser gjør. En utfordring vil da være å bygge inn nødvendige mekanismer for kontroll av signalene, mer enn standard ATM management. Kanskje kan man også se for seg miksing, editering, grafikk- og lydbehandling i samme enhet, bygget over en ATM-kjerne. Tilpasning av forskjellige videoformater vil måtte gjøres med adapterkort.

For å trekke det enda lenger kan man også vurdere digitale videosignaler på IP-format. Utfordringen vil da bli å bygge tilsvarende kontrollmuligheter inn i enhet basert på en IP-ruter.

7.3 Area(s) of focus

Noen interessante områder i forbindelse med streaming av video i heterogene nett vil være innenfor testing og prototyping. ATM er en mer moden teknologi enn IP når det gjelder tjenestekvalitet, og kvalitetsmekanismene er veldefinerte. For IP derimot, foregår det et aktivt standardiseringsarbeid, men det er ennå et stykke igjen til mekanismene er på plass. Å teste ut heterogene nett som nevnt i foranstående kapittel bør være et hovedpoeng her.

For produksjonsdelen er det allerede kommet ATM svitsjeprodukter på markedet. Det vil være viktig å se på muligheter for (eller nødvendigheten av) å produsere ATM-utstyr.

Når det gjelder IP i produksjonsfasen er det et stykke frem til dette blir konkurrerende teknologi til ATM, men det bør holdes et øye med hva som skjer.

References

[1] "ATM Pocket Guide", Wandel & Goltermann, http://www.wg.com

[2] "White Paper - QoS protocols & architectures", Stardust.com, skrevet for QoS Forum, http://www.stardust.com, http://www.qosforum.com

8. Distributed Video Production

8.1 Description of the topic

Distributed video production refers to situations where the cameras, recorders, switches, mixers and other equipment used in video production (or post-production) are located at several sites linked by high bandwidth network connections (see Figure 2). The DVP project will investigate user requirements for several forms of distributed video production and will run a series of trials of a distributed virtual studio, a distributed rehearsal system and a distributed video editing and retrieval system.

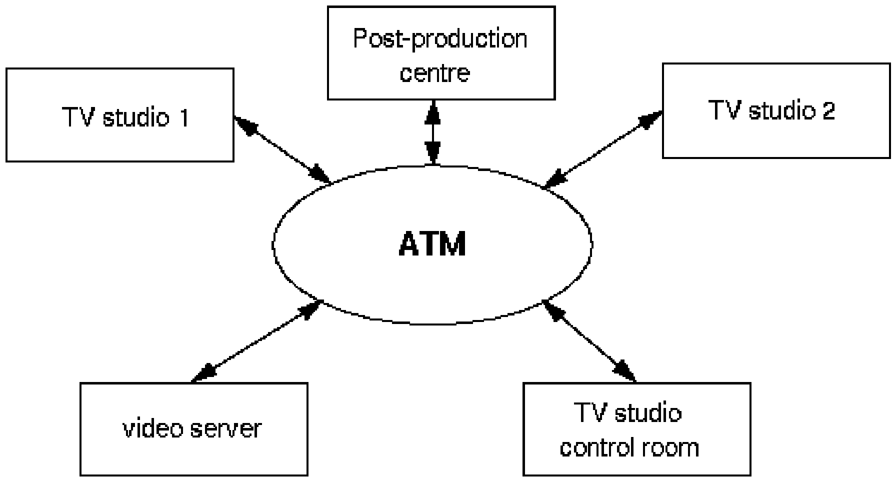

Figure 2: Distributed Video Production

8.2 Major issues and challenges

MPEG II compression has been designed so far mainly for program distribution at relatively low data rates (8 mbs) per stream. There will be a new MPEG II variant for professional use which can be better suited for the video production chain (contribution, production, post- production and distribution). But there is still an open question how compression will influence the video quality if compressed video is used and concatenated in the different production stages.

On the other hand, the usage of compression offers the choice to get video transferred online to places, which could not be interconnected before. New transport layers like ATM imply that the video signal will no more be carried in continuous streams, but will be broken into pieces and packetized into cells. The characteristics of the underlying transfer system are quite different than with isochronous D1 streams, quality of service aspects like cell loss and jitter will be important. But packetization offers the chance that video distribution is no longer limited to local distribution.

In detail, questions like the following ones have to be answered:

- How can ATM-connectivity based on SDH (STM1 to STM4) be used for connecting CCIR 601 studio systems?

- Transmission and processing delays: what is the impact of such delays during video pro- duction, what are acceptable delays, where is isochronous transmission really required?
- Synchronization: what are the synchronization requirements for tightly-coupled dis- tributed video processing?
- What quality of service of the transmission system is required?

8.3 Related Work

A brief presentation of *The DVP project (Distributed Video Production)*, ACTS project AC089 (Q4 1995 – Q1 1998), is provided amongst the **Reference Materials** in this report (see section 21).

9. HDTV in Internet2

This is a research project conducted on the Internet2 network. A short introduction taken from home pages of the project is given here, a more complete documentation is given in the **Reference Materials** in this report (see section 23).

9.1 Overview

Engineers at the University of Washington, working with colleagues at Stanford University and Sony Electronics, have developed technology to send studio-quality high definition television signals over the Internet. The demonstration project, developed in support of the ResearchTV consortium, differs from previous efforts to send HDTV over data networks in that it does not utilize dedicated circuits or ATM network technology. Rather, this effort relies exclusively on Internet technology and the capacity of the "Abilene" Internet2 research network.

9.2 Background

In February 1999 ResearchTV and the University of Washington conducted one of five high-bandwidth experiments selected by Internet2 developers to successfully test the coast to coast connectivity of "Abilene," an Internet2 backbone network. In that case, the goal was to send 7 Mbps MPEG-2 (standard definition broadcast quality) research video streams from the University of Washington in Seattle to Union Station in Washington, D.C., the site of Internet2's inaugural celebration.

This latest demonstration pushes the performance frontiers of the Internet2 research network even further by sending high definition television streams at 40 Mbps and 270 Mbps. This project will give Internet2 developers crucial information about the behavior of the network under heavy load, as well as opening new frontiers for the broadcast industry. This is consistent with the primary objective of Internet2: to enable development of advanced applications that would not be possible without the capabilities of the Internet2 Abilene backbone network.

9.3 Goal

The project goal is to demonstrate the feasibility of sending continuous streams of broadcast-studio-quality high definition (HD) video over a general purpose, multi-user IP network. In particular, from Stanford University to University of Washington, via Internet2's Abilene backbone, while maintaining perfect high definition picture quality.

Notably, for this experiment, the Internet2 capacity will not be reserved or pre-allocated, nor will any Quality-of-Service (QoS) or packet prioritization mechanisms be used within the network. That is, the HD video packets will be contending for bandwidth along with other Internet2 traffic. Beyond proving the feasibility of using only Internet technology for HD video transmission, it is also a goal to understand the effect of high-bandwidth applications on other traffic, and vice versa.

The project encompasses two distinct efforts, both based on broadcast industry standards. First is to send a 40 Mbps DVB-ASI stream; second is the more ambitious goal of sending a Sony HDCAM®/SDTI stream at 270 Mbps.

References

[3] http://www.internet2.edu/html/news.html

[4] http://apps.internet2.edu/talks/

10. Synchronisation of SFN in IP Networks

10.1 Description of the topic

Protection - the disaster Scenario

Single Frequency Networks (SFN) accept in the order of 1 sec. interruption of signal between signal entry point and transmitter. This requirement comes from the use of GPS as synchronisation source for the network where GPS deliver the high stability 10 Mhz and 1 Hz pulse to the network.

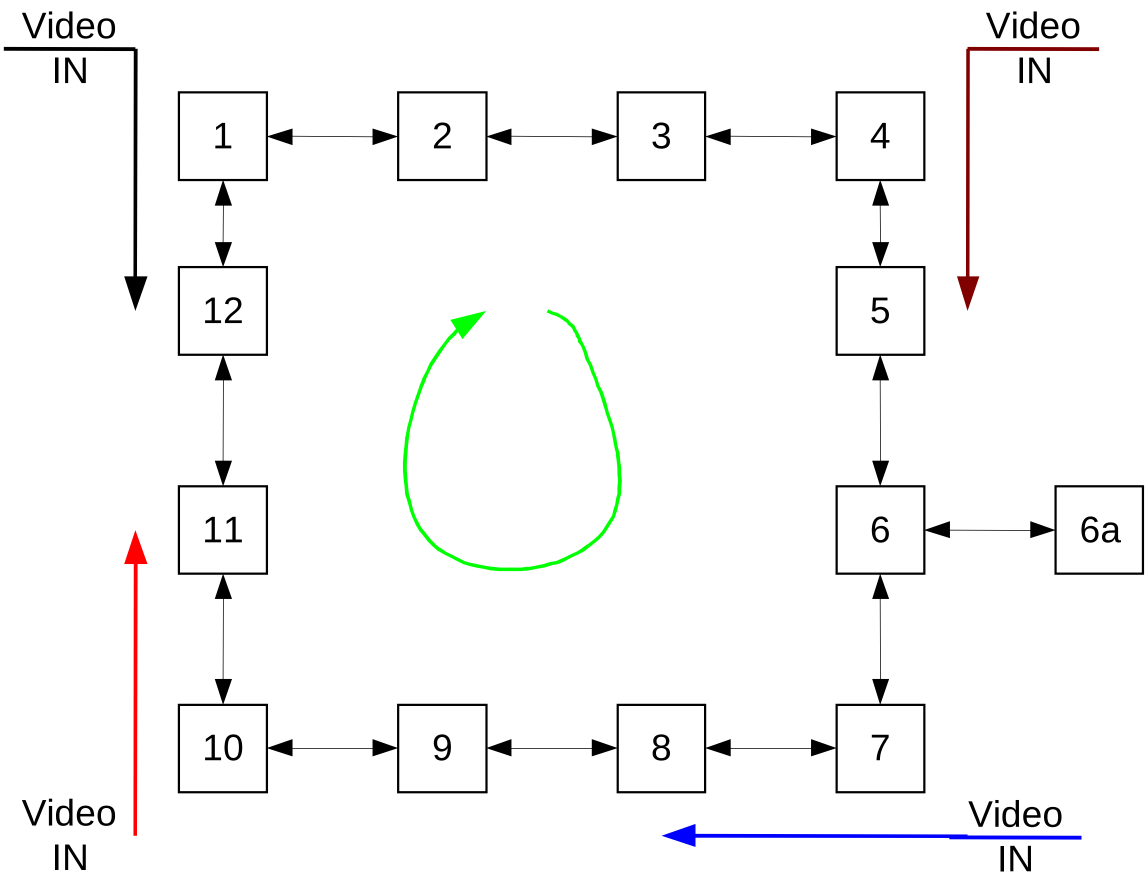

Figure 3:Typical distribution network

Figure 3 show a typical distribution network based on transmitter nodes 1 - 12 where the signal is transmitted in a ring configuration to all of the transmitter sites. The input signal can be introduced a different nodes in the network depending on where the video production takes place. In case of a signal break two solutions can be used to achieve continued signal delivery:

- The signal path between signal entry and transmitter can in case of a fault be redirected/rerouted by establishing a path in the opposite direction of the ring.
- The signal can be transmitter in two directions in the ring, thus enabling the receiver to choose the best quality signal to be used in the transmitter.

The first solution include that the established signal path can be re-established within the time frame of 1 sec. for SFN. This include not only redirection of signal in the distribution network, but also resynchronisation of DVB MPEG-2 coder and decoder and network adapters.

The second solution is that the transmitter receives two data streams and can choose between the streams. This is however a more expensive solution and requires duplicated network capasity.

Normal requirement is that its not acceptable to have a break of more than a few milliseconds in the TV transmitted signal.

10.2 Major issues and challenges

10.2.1 Synchronisation - different network technology

Below is given some considerations regarding recovery times for components and network technology.

- Re-establishing of the SDH trail takes in the order of 1 - 10 second if the trail has to be restored by setting up a new trail by switching commands to each network element. If path protection is applied where the signal is transmitter over two independent parts of the network, the switchover time is less than 50 ms
- Re-establishing of the ATM connection can be done within X seconds depending of the size of the netowork.
- Resynchronisation of the Network Adapter - delay X seconds
- Resynchronisation of the Video Codecs / DVB-Encoder - X seconds.

10.2.2 Synchronisation - a chained process

When the SFN network is in sync and a fault occur, the individual element has some buffer capacity to compensate for disruption of the input signal. The distribution in SFN have a layered structure of equipment that in permanent sync to deliver a DVB signal to transmitter. The layered structure can be as below from top and down.

- Single Frequency network (SFN) synchronisation
- Video codec synchronisation
- Network adapter synchronisation
- ATM network synchronisation
- SDH network synchronisation

As an example if SFN transmission is based upon ATM over SDH, a disruption of the SDH signal may trigger a ATM path reconfiguration and consequently if the reconfiguration delay trigger a network adapter re-sync then the consequence can be that the complete SFN network is out. When the chain fails, everything must be reconnected again.

Time, until the Videosignal is back will depend of the re-synchronisation delay in each component in the transmitter chain and the interactions between the individual components. More work are needed to understand the interaction between the technologies in a SFN distribution network.

10.2.3 Synchronisation - with IP technology

Issues related to IP transmission. The assumption is that IP substitute ATM and that Packet Over Sonet (POS) can be used. The following functionality in IP networks need to be better understood before introduced in SFN networks.

- QoS - Guaranteed delivery
- MPLS and tunnelling
- Connection Less Network Layer Routing
- TCP transport streams

11. Multi Protocol Label Switching

11.1 Description of the topic

The shortcomings of the Internet's routing methods have become apparent with the growth of the Internet and increasing functionality in IP. Best-effort service and routing via the shortest path (the traditional characteristics of IP networks) are no longer sufficient.

For IP-based voice and video, service providers and IT managers need the ability to route timing-sensitive traffic over low latency paths. When congestion occurs on a primary network path, operators need a way to shift some of that traffic to a parallel link, or to dump traffic to a preferred alternate route in the case of network failure. None of this is possible with the current IP-routing mechnism.

Currently, routers and other layer-3 devices forward IP traffic based on matching the IP address in a packet with an entry in the routing table of a layer-3 device. Routers scan the packet header until they arrive at the longest prefix match. MPLS uses label swapping or switching for traffic forwarding and allows streams of data to be forwarded based on an exact match with a short, fixed-length label. The labels can also be used to identify classes of traffic, so that streams of data can be forwarded identically.

IETF has completed work on the core standards for MPLS, including the mechanisms that network operators need to set up explicit paths through a network (vs. hop-by-hop shortest paths). MPLS is in the introduction stage, and is expected to mature over the next year. The protocol implementations are ready and some major operators will start testing of MPLS. Products that employ MPLS are just beginning to hit the market, with many more to come in the second half of 1999 and the early part of 2000.

11.2 Major issues and challenges

The QoS/CoS aspects of MPLS, the traffic engineering capabilities of MPLS and the concept of tunneling IP packages through the network, will be of advantage in video distribution and contribution.

There are some outstanding issues that need further attention:

- MPLS in combination with ATM need further study,
- MPLS in IP networks is dependant of interoperability between vendors,
- Multicast is currently not included in MPLS standards,
- MPLS tunnelling of video streams in relation to QoS need further study.

The article ”IP and Multi Protocol Label Switching” provides a further look into MPLS. An extract from this document is included amongst the **Reference Materials** in this report (see section 20).

References

[5] Petrosky, M., “IP and Multi Protocol Label Switching”, from the column *Performance Computing*, in Internet Packets & Protocols, 1998/1999.

12. Advantages for IPv6 in relation to DVB and DVD

12.1 Description of the topic

Multicast is already available in IPv4, and IPv4 has to some extend the necessary functionality for multicast distribution of video. Multicast applications have been developed for IPv4, but IPv6 extends IP multicasting capabilities by defining a much larger multicast address space. All IPv6 hosts and routers are required to support multicast. In fact, IPv6 has no broadcast address as such; it has various multicast addresses of various scopes. The improved scoping offered in IPv6 promises to simplify the use and administration of multicast in many applications.

The extended header field for QoS (flow label field) gives a more efficient access and reading of the packet header compared to IPv4, when the router need to classify the individual packet streams (Class Of Service - CoS).

IPv6 has mandatory support of security and mobility (this is not the case for IPv4). Distribution of secure packet streams is made easier, however all routers in the chain must support IPv6.

12.2 Major issues and challenges

The document ”The Case for IPv6” [6] — a draft document from IAB Working Group of the Internet Engineering Task Force (IETF) — describes briefly the technical and commercial advantages of IPv6 compared with the existing IPv4 solutions. An abbreviated version of this document is included amongst the **Reference Materials** in this report (see section 19).

The main headlines are:

- Extended IP address space
- Security protocol definitions
- Mobility for IP terminals
- Network Administration
- Extended meader fields for QoS

The primary issues in this project concern how the new header features in IPv6 might be used effectually within the contexts of DVB and DVD.

References

[6] King, S., Fax, R., Haskin, D., Ling, W., Meegan, T., Fink, R., Perkins, E., ”The Case for IPv6”, IETF Draft draft-ietf-iab-case-for-ipv6-05.txt, 22 October 99.

13. Network Management

13.1 Description of the topic

With the new digital technologies countrywide video contribution and distribution broadcasting networks are increasing more and more in it's complexity. Together with the Multi-Service Network approach from modern ATM networks, the merging of the different services are very attractive to broadcast network carrier. They can combine video contribution and distribution, together with data and Telephony traffic. The means a normal broadcast network can exists of the following elements:

- ATM Switches
- IP Router – Core
- IP Router – Access
- Network Adapter
- Video Codec
- Video Switches
- DVB Transmission Equipment
- Radio Relay Equipment
- PBX – Private Branch Exchange (ex. Hicoms, plus Voice over IP)
- IT-Server

For interactivity TV is also necessary to integrate return channel technologies like

- POTS
- ISDN
- xDSL

The TV program content provider has also taken into account. For additional services wireless ATM, Statistics and billing could also be topics for the future.

13.2 Major issues and challenges

13.2.1 NMS Structure

The primary question in this area concerns how to manage such a complex network. Specifically, the question of NMS structure is also a question of is it really necessary to bring all technologies under one NMS together and what is the role of each network user? These can be separated into different groups:

- The network carrier (can be TV station also)
- The broadcaster (TV stations, post production companies)
- The content provider
- Telecommunication service provider

Spitting the network equipment into these groups it is very difficult to integrate all equipment and their different management solutions into one system. So there have to be a middle way of integrating and realising an overall NMS.

13.2.2 Two different NMS concepts

There are two different concepts thinkable which has to be investigate in details: single NMS concept vs. separated NMS concept.

Single NMS concept

It must be investigated what is possible, what is usable, what is realistic. Based on MainstreetXpress products, the concept could be as shown in Figure 4.

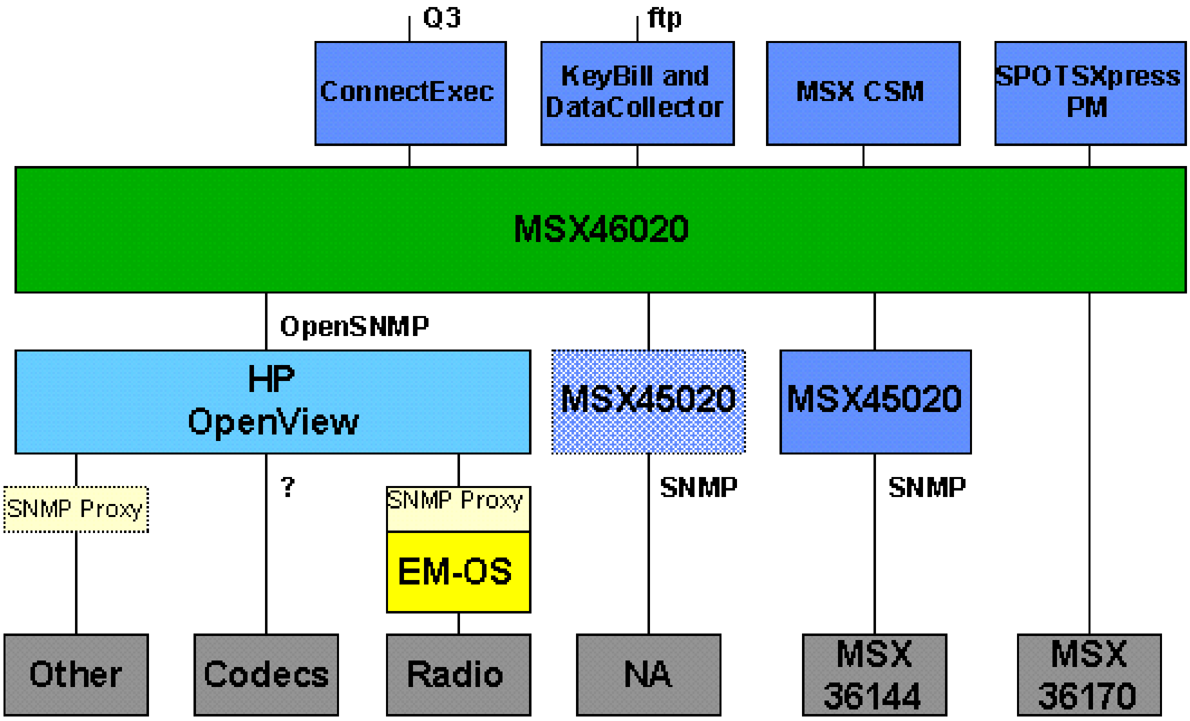

Figure 4

Separated NMS concept

This concept assumes that it is possible to split the network into the network and video equipment parts. This means that there are two NMS:

- Network management station
- Video network management station

If it is possible to concentrate at least the alarming etc. in one overall NMS must be investigated further on.

14. Influences of Radio Relay parameters upon ATM QoS and video quality

14.1 Description of the topic

Regarding modern carrier networks, fiber is used to have a saving of investments for maximum bandwidth and to be open for future technologies like Wave Division Multiplexing (WDM). Normally no major problems could occur and weather condition influences won't happen.

Regarding TV distribution networks, Radio Relay is the preferred technology to link a country wide TV Transmitter systems together. The disadvantage is that under special weather conditions, Radio Relay connections can be interrupted and it is interesting, what are the requirements to the involved equipment to cover with these interruptions with synchronisation and data error.

14.2 Major issues and challenges

In this area, the following issues are worthy of investigation::

- Influences of Multipath Fading (BER) to the ATM QoS.
- Influences of Multipath Fading (BER) to IP connections.
- Jitter and wander of Radio Relay and the incluences to Cell Delay variation (CDV).
- CDV and Network Adapter (NA). Is the only solution to increase the NA buffer and to cause bigger delays?
- What is the balance between Forward Error Correction overhead (Reed Solomon) and what it is really necessary. How good is Read Solomon to cover Radio Relay Bit Error Rates.
- What is the limit of the combined MPEG TS recovery system in combination NA and video decoder? What can standard ATM offer?
- What about resynchronisation times of the synchronisation chain, when the link is interupted.
- What relation is between Radio Relay BER and CDV ?

15. Digital audio in professional applications

15.1 Description of the topic

Today, compressed digital audio has gained popularity in many applications. A lot of work is being done regarding transportation of compressed audio over general data networks, both as audio files and as bit streams in real time. Often, compressed audio is also handled as a part of a digital video signal. EBU and SMPTE have delivered a comprehensive report regarding exchange of programme material [1], with focus on video.

Compressed audio is of course attractive, especially for transmission over long distances and computer storage. There are, however, at least three disadvantages for using compressed audio in high quality studio applications:

- Cannot, in most cases, directly be mixed or manipulated.
- Almost all compression algorithms introduces delay.
- Almost all compression algorithms introduces artifacts which may be audible in certain circumstances.

For studio and other high-quality applications, linearly encoded audio therefore will be used also in the future. Two standards is currently in widespread use for transport and interfacing linearly represented audio, AES 3 for two-channel and AES 10 for multichannel audio. Additionally, a number of proprietary formats are used.

15.2 Report from the Lawo ATM Symposium –99

In this document is included a report entitled *”High quality Audio over ATM – Report from the Lawo ATM Symposium –99”*. This abstract from this report reads:

> *Only two standards do currently exist for transfer of professional, linear coded audio. One for two-channel audio, and one for 56 channels. Both are point-to-point connections.*
>
> *There are therefore a demand for standards for transferring audio over more general networks. The ATM network seems promising because defined Quality of Service is a part of the protocol, and the inherent delay is low. Streaming audio is very critical regarding delay, and especially variations in delay.*
>
> *One company, Lawo, has already started manufacturing equipment based on ATM. But because they know a standard is the basis for success, they invited a number of competitors to a symposium for discussing a possible standard for audio over ATM.*
>
> *The participants agreed on a basic format for transferring two-channel audio in a virtual connection. But a lot of details still remain. AES has also started some activities in this field.*

The report provides an in-depth look into the latest work in the area of professional digital audio, and can be found amongst the **Reference Materials** in this report (see section 22).

16. Krav til Forsinkelse og Bildekvalitet

16.1 Synkronisering/forsinkelse

16.1.1 Program produksjon:

- Direkte-sendte program typer: Kultur, nyheter, fakta, underholdning, undervisning.

  Signaler som benyttes til direkte programoverføringer eller opptak, må være synkrone. Dette er svært viktig om man ikke skal få rivninger i bildet under svitsjing mellom forskjellige programkilder. Svitsjing skjer i linje 6 eller linje 319. Programkilder som kommer utenfra må være ”real time”, men behøver nødvendigvis ikke være synkront med ”Timing Reference Signal”(TRS) for programavviklings stedet, men må likevel ikke være forsinket mer enn 40ms (en hel ”frame”) for at det skal være mulig å synke signalet opp mot systemet på avviklings stedet.

- Etterbearbeidete program typer: Drama, film, nyhetsinnslag, fakta, viten, underholdning, kultur.

  Ved editering i en ”post production” lokalt må signalene være synkrone. I et TV-hus finner en vanligvis at et hovedsynkroniseringssignal blir distribuert til flere lokale studioer og editeringsrom. Er en avhengig av å hente programmateriale fra et sentralt sted og arbeider i ”real time”, må editerings plassen være synkron med det ”sentrale” signalet.

16.1.2 Lyd & bilde

- Lyd og bilde blir produsert separat før det blir et ferdig programmateriale. Kravet til det ferdige programmaterialet er at lyd og bilde er i synk (lip sync). Det samme vil gjelde for signaler som kommer utenfor produksjonstedet der lyd og bilde kommer samlet inn til stedet hvor programmet avvikles. Forsinkelse mellom video og audio signaler i et digitalt studio kan forsinkes max.60ms.

  I nyhetsprogrammer er det viktig at forsinkelse i signalet ikke oppstår mellom intervjuer/reporter i felten og studio, da det kan skape forvirring mellom intervjuer/reporter og nyhetsoppleser.

- Programutveksling mellom TVstasjoner som sendes direkte må også være synkrone med hensyn til både lyd og bilde.

16.1.3 Overspilling

- Programmateriale for kun overspilling vil ikke ha krav til synkronisering med andre programkilder. Overspilling kan foregå over timer om det er ønskelig med hensyn til tid og pris. Det er kun overføring i produksjonsammenheng det er krav til kontinuerlig overføring i ”real time”.
- I nyhetssammenheng kan det være aktuelt å overspille hurtigere enn ”real time” da dette kan bety mye med hensyn til tiden.

16.2 Bildekvalitet

- De fleste programtyper i programproduksjon setter høye krav til bildekvalitet. Typiske programformer: drama, film, fakta, kultur, viten, undervisning.

  Video trenger stor båndbredde for å kunne gjengi god bildekvalitet. For digital video i programproduksjon er data-raten 270Mb/s, SDI signal. Dette signalet er full-format video og er ikke komprimert. Sålenge et videosignal skal prosesseres/mixes og chroma-key benyttes må en arbeide med full-format. Komprimerte signaler på 2:1 kan skape vanskeligheter og må veien om full-format. Kodingen må være >50Mb/s. SDI muliggjør også filoverføring av protokoller på en 270Mb/s ”carrier”.

- Programtype: Nyheter.

  Nyheter med innslag fra fjernt og nær kan aksepteres med redusert bildekvalitet. En setter ikke så høye krav til bildekvaliteten i innslag som har nyhetens interesse. Det samme gjelder programformer med innslag fra amatøropptak, eller andre type medier med lavere bildekvalitet.

- 360Mb/s tyder også på å bli standard for TVproduksjon i fremtiden.

  Problemstilling med bitreduksjon:

- Bildekvaliteten som oppnås i systemer hvor bitrate reduksjon (BBR) brukes avhenger av strukturen i det aktive bildet og vertikal blanking intervallet. Bruken av BBR kan ødelegge test signalene (ITS) og data signalene i vertikal blanking intervallet, og noen spesielle test signaler som ligger i det aktive bilde området.

Følgende standarder som brukes: 140Mb/s (G703), 34-45Mb/s (G703), MPEG-2.

Standardene er typiske Telecom standarder. TV-selskapene forholder seg til disse (MPEG, DVB).

Krav som TV selskapene vil stille til distribusjon av sine DVBsignaler, er at innholdet er urørt fram til mottaker.

- Det er vanskelig å kvalitetsbedømme om et bilde er godt nok, fordi det avhenger av om signalet skal til en sluttbruker, eller om signalet skal inn i produksjon igjen. Pr. idag må en vite på forhånd hva bruken er.
- EBU Technical Review er et hjelpedokument med retningslinjer rettet mot fabrikanter slik at produkter får et interface/standarder som kan ”snakke” sammen på tvers av fabrikanttyper.

16.3 Utfordringer

- Finne/sette om mulig en standard for hvor god/dårlig bildekvaliteten kan være ved:

  DVproduksjon

  DV utveksling av programmateriale

  DV distribusjon

References

[7] Tech 3283, 2.2.6 Switchingpoint of a video switching matrix

[8] Pflichtenheft Nr. 8/1.1

[9] Tech 3283, 1.3 Other standards to be considered

17. Tjenestekvalitet ved distribusjon av digital video over IP/ATM nettverk

17.1 Description of the topic

Distribusjon av digital video kan foregå enten som filoverføring eller streaming. Ved filoverføring flyttes eller kopieres en videofil slik at det lages en nøyaktig kopi av den opprinnelige filen hos mottaker. Retransmisjon av enkeltbiter ved mulig pakketap er derfor nødvendig. Det er ingen krav om synkronisering eller bestemt bitrate ved overføringen, men datamengden medfører at nettverkskapasiteten vanligvis er stor. Ved streaming overføres og avspilles en videostrøm i sann tid til mottaker. Det foregår derfor ingen form for retransmisjon av korrupte data, eller unødvendig bufring hos mottaker. Derimot eksisterer det strenge krav til forsinkelse, synkronisering og overføringsrate for at kvaliteten på avspillingen skal bli tilfredsstillende.

Distribusjon av digital video er viktig både i kontribusjon og ren distribusjon. Ved kontribusjon menes distribusjon av programelementer for digital video med formål å sette sammen disse til programmer. Kontribusjon foregår mellom produsenter av slike programelementer, gjerne lokalt i et produksjonsselskap. Dette stiller strenge krav til at bildekvaliteten beholdes, og programelementene komprimeres derfor ofte ikke. Ved ren distribusjon menes overføring av ferdige programmer mellom kringkastere og TV-sendere. Slik distribusjon foregår ofte over lange avstander og den digitale videoen komprimeres derfor gjerne for å spare båndbredde.

Denne tekniske oppsummeringen begrenses til streamingbasert distribusjon av komprimert digital video.

Enhver netttverksløsning av slik distribusjon forutsetter at ønsket kvalitet ved bruk av tjenesten lar seg spesifisere. Tilgjengelige nettverksressurser er jo alltid begrenset, så ressursdeling er en nødvendighet. To fremtidige kandidater er bruk av dedikerte IP eller ATM nettverk. Hovedpoenget med denne oppsummeringen er å beskrive dagens muligheter for spesifikasjon av tjenestekvalitet ved disse nettverksløsningene. Fokus vil være på protokoller på transport- og nettverkslaget, ikke på selve transportsystemet.

17.1.1 IP nettverk

Først en presisering. Med IP nettverk i denne forbindelse menes først og fremst dedikerte leide linjer eller VPN, som er basert på IP-teknologi, men som er lukket for annen trafikk. Distribusjon av digital video sammen med Internettrafikk behandles ikke i særlig grad da dette ennå ligger lengre fram i tid enn dedikerte nett.

IP tilbyr i utgangspunktet ingen mekanismer for tjenestekvalitet. Hver IP-pakke er en selvstendig enhet, som blir behandlet likt av nettverket (med unntak av kontrollpakker). Den eneste grunnleggende tjenesteklassen er derfor best-effort.

En tilsynelatende enkel løsning er derfor å sørge for at nettverket alltid har (mer enn god) nok kapasitet slik at kvalitetskravene til streaming alltid kan tilfredsstilles. For distribusjon av digital video synes dette lite realistisk, grunnet de store datamengdene som er involvert. Mekansimer for tjenestekvalitet er derfor nødvendig.

Muligheter og mekanismer

Kravene til streaming av digital video gjør at transportprotokollen må være UDP, da retransmisjon (og dermed TCP) ikke tillates. UDP er en forbindelsesløs og upålitelig transportprotokoll. Det eksisterer heller ikke noe tidsbegrep i IP, så for synkronisering er bruk av RTP en nødvendighet. RTP tilbyr enkel tidsstempling av pakker og mekanismer for å formidle kontrollstatistikk.

I de seneste årene har det vært arbeidet hardt i IP-verdenen for å utvide dagens arkitektur med alternative tjenesteklasser. Integrated Services (IntServ) fra IETF [14] er det viktigste arbeidet i så måte. Arbeidet har resultert i to nye tjenesteklasser, "Guaranteed" og "Controlled-load". Begge er basert på en strømmodell ("flow") av trafikken, og benytter en token bucket modell ved spesifikasjon av trafikkparametre. "Guaranteed" [17] tilbyr harde garantier ("upper bound") på parametre som maksimal forsinkelse, og overføringsrate, under forutsetning av at trafikken følger en overordnet fluidmodell. "Controlled load" [16] tilbyr en mykere garanti, karakterisert ved en oppførsel som et lett belastet best-effort nettverk, også ved trafikkopphopning.

IntServ forutsetter videre at en signaleringsprotokoll benyttes for å formidle trafikkspesifikasjonene mellom ruterne i nettet. Den vanligste protokollen til dette formålet er RSVP [15], som er en "soft state", mottaker-initiert reservasjonsprotokoll. Bruk av RSVP er mulig både for unicast og multicast forbindelser.

Kombinasjonen av disse mekanismene implementert i rutere og vertsmaskiner muliggjør ende til ende garantier over IP nettverk. Den store ulempen er imidlertid at dette skalerer dårlig i store nettverk, grunnet mye ekstra overhead, som f.eks. tilstandsinformasjon i ruterne. For små nettverk (få noder) er imidlertid ikke dette like kritisk.

For å bøte på bl.a. skaleringsproblemet har det også vært arbeidet med en enklere modell for differensiering av trafikk innen IETF, kalt Differensiated Services (DiffServ) [18]. Denne utvidelsen antas å være mer velegnet på Internett, men er fremdeles under utprøving. DiffServ tilbyr ingen globale garantier, kun (lokale) per hop garantier, med enkel trafikkprioritering basert på ToS-feltet i IP headeren.

Som tidligere nevnt er fokus for denne oppsummeringen mekanismer på transport- og nettverkslaget. Alle mekansimene over må imidlertid mappes til underliggende transportsystem. Det er utarbeidet spesifikasjoner for hvordan dette skal skje både over Ethernet og ATM, men disse behandles ikke her. Se [10] for en gjennomgang av de viktigste.

Dagens praksis

Generelt er bruken av mekanismene for tjenestekvalitet over IP nettverk ennå i en tidlig fase. Man har så smått begynt å få noe erfaring med bruk av IntServ sammen med RSVP. Spesielt gjelder dette innen dedikerte nettverk. Skaleringsproblemet har imidlertid bremset utbredelsen noe. Det finnes stabile implementasjoner for de fleste rutere og vertsmaskiner.

Erfaringene med DiffServ, og ikke minst mulige tjenester basert på en slik trafikkprioritering er enda lenger unna. De tidlige forsøkene gjøres over et virtuelt testnettverk, Qbone. Men det arbeides intenst og interessen blant nettoperatører er stor. Applikasjonsområdet for DiffServ er imidlertid mer IP telefoni, enn streaming av digital video.

Når det gjelder distribusjon av digital video sammen med annen Internettrafikk har det første forsøket på Internet2 blitt foretatt med overføring av 40Mbps og 270 Mbps MPEG2 strømmer. Ingen reservasjon eller prioritering av trafikken, og heller ingen

ATM teknologi ble benyttet (kun Gigabit/Fast Ethernet), se kapittel 23.

Det er også utarbeidet en ny Internet Protocol (IPv6), som på sikt er tenkt å erstatte dagens IPv4. Støtten for tjenestekvalitet er tilnærmet identisk. Det eneste nye er inkludering av et felt for identifikasjon av strømmer ("flow label"). Den nøyaktige bruken av dette feltet er imidlertid ennå ikke fastlagt (se kapittel 12).

17.1.2 ATM nettverk

En presisering er nødvendig også her. Med ATM nettverk i denne forbindelse menes nettverk hvor kun mekanismer i ATM benyttes på transport- og nettverkslaget. ATM som ren link-lags teknologi under f.eks. IP behandles ikke, da en slik bruk gjerne forutsetter permanente forbindelser og avhenger av mulighetene for å spesifisere tjenestekvalitet på lagene over.

ATM er en meget kompleks teknologi, og spesifikasjon av tjenestekvalitet kan gjøres på flere nivåer avhengig av trafikktype. Denne oppsummeringen begrenses til det som er interessant for distribusjon av digital video, hvilket forutsetter en forbindelsesorientert og sanntidsavhengig overføring. Se [11] for en generell oversikt over ATM.

Muligheter og mekanismer

De aktuelle tjenesteklassene i ATM ved streaming av digital video er enten CBR eller rt-VBR. CBR tilbyr en konstant bitrate med tidsreferanse for synkronisering, mens rt-VBR tilbyr en variabel bitrate, også med tidsreferanse, hvilket gjør den mer effektiv hvis bitstrømmen ikke varierer.

ATM-forbindelser er enten permanente og forhånskonfigurerte (PVC) eller opprettes dynamisk ved behov (SVC). For hver forbindelse opprettes en trafikkontrakt som spesifiserer trafikken.

En trafikkontrakt karakteriseres av tjenestetype (sett av trafikkparametre) og tjenestekvalitet (sett av QoS parametre). For CBR er dette henholdsvis PCR og CDTV peak, samt CLR, CDV, og mean CTD. For rt-VBR er trafikkparametrene PCR, CDTV, SCR og MBS, mens QoS parametrene er CLR, CDV, og max CTD. Parametrene er enten forhåndsdefinert av nettoperatøren, eller kan spesifiseres av brukeren [12]. Alle parametrene behøver ikke å bli spesifisert for å definere en gyldig trafikkontrakt.

I ATM er det også definert ulike QoS-klasser som er uavhengige av selve tjenesten. CBR tilhører klasse 1, mens VBR tilhører klasse 2. Klassene differensieres ved forskjellige verdier på QoS parametrene.

Et antall funksjoner for traffikkmanagement i ATM sørger for at trafikkontraktene overholdes og nettverket ikke blir overbelastet. Trafikkforming er et eksempel på en slik funksjon.

Dagens praksis

Generelt er bruken av mekanismene for tjenestekvalitet i ATM godt utbredt. Mange nettoperatører tilbyr flere av de ulike tjenesteklassene, og lar brukerne spesifisere mange av parametrene i trafikkontraktene før forbindelsen settes opp. Det er imidlertid stort sett alltid snakk om permanente forbindelser (PVC). Dynamisk oppsatte forbindelser (SVC) tilbys sjelden.

Bruken av ATM ved distribusjon av digital video er allerede blitt demonstrert i flere sammenhenger. Et eksempel er NRK som benyttet ATM nettverk ved deler av distribusjonen under OL 1994 på Lillehammer, og med godt resultat. Videre anbefaler EBU at ATM nettverk benyttes ved streaming både i lokalnettverk (LAN) og wide-area nettverk (WAN) [13].

Imidlertid har Europa og USA valgt forskjellige tilnærminger når det gjelder valg av tjenesteklasse og adapsjonslag ved distribusjon av video over ATM. I Europa satses det på CBR og AAL1, mens USA går for rt-VBR og AAL5.

17.2 Major issues and challenges

Utfordringene når det gjelder spesifikasjon av tjenestekvalitet ved streaming-basert distribusjon av digital video er forskjellige for IP og ATM nettverk.

For ATM nettverk er mekanismene allerede på plass. Veldefinerte trafikk- og QoS-parametre eksisterer. Ulike tjenesteklasser tilbys av nettoperatører. Teknologien regnes som moden, og er allerede benyttet i noen tilfeller av distribusjon av digital video. I tillegg har de første nettverksadaptere med ATM interface begynt å dukke opp på markedet. Utfordringene er derfor av mer generell karakter enn direkte relatert til spesifikasjon av tjenestekvalitet :

- Hvilken kombinasjon av tjenesteklasse og adapsjonslag (og trafikkontrakt) er best egnet for selve distribusjonen?
- Hvordan håndtere f.eks. re-synkronisering ved kritiske situasjoner hvor videostrømmen plutselig uteblir? (se kapittel 10)
- Er det spesielle forhold som influerer på spesifisert QoS, f.eks. bruk av radiotransmittere? (se kapittel 14).

For IP nettverk derimot er situasjonen en annen. Få tjenesteklasser er definert. Mekanismene for å overholde tidskritiske parametre er enten ikke fullt utviklet, eller trenger å testes nærmere i full-skala omgivelser. Dette gjelder spesielt "Guaranteed" tjenesteklasse i IntServ, som er det nærmeste man kommer CBR/rt-VBR i ATM. Utfordringene er derfor fremdeles tilstede når det gjelder å få selve rammeverket og kontrollmulighetene på plass.

Et åpent spørsmål er også hvor viktig spesifikasjon av tjenestekvalitet er for returkanalen, f.eks. i nettbaserte applikasjoner som nettauksjoner. Og vil likevel det å alltid sørge for overkapasitet i det dedikerte IP nettverket holde? Ihvertfall så lenge det ikke er snakk om integrering av andre typer trafikk sammen med videostrømmen.

Kort oppsummert er utfordringene ved IP nettverk av mer grunnleggende karakter enn ved ATM nettverk.

Referanser

[10] Ferguson P. & Huston G., "Quality of Service - Delivering QoS on the Internet and in Corporate Networks", 1998

[11] "ATM Pocket Guide", Wandel & Goltermann

[12] "Produktbeskrivelse Nordicom ATM", 1.10, 1999

[13] "Final Report: Analyses and Results", EBU / SMPTE Task Force for Harmonized Standards for the Exchange of Programme Material as Bitstreams, 1998

[14] "Integrated Services in the Internet Architecture: an Overview", RFC 1633, 1994

[15] "Resource ReSerVation Protocol (RSVP) -- Version 1 Functional Specification",

RFC 2205, 1997

[16] "Specification of the Controlled-Load Network Element Service", RFC 2211, 1997

[17] "Specification of Guaranteed Quality of Service", RFC 2212, 1997

[18] "An Architecture for Differentiated Services", RFC 2475, 1998

18. Interactive Television

18.1 Description of the topic

Interactive Television (ITV) is a technological possibility which is well on its way to becoming a reality for the common consumer in the industrialized world. A great number of both large and small industrial actors are already developing and deploying ITV, in order to best position themselves for what can potentially become an enormous source of revenue: the potential target group for ITV is *any* television viewer.

Precisely speaking, ITV is not synonymous with *Digital TV*; however, it is clear that the possibilities for ITV will be much richer when founded upon Digital TV. A narrow definition of Digital TV could simply imply that the signals sent from a broadcaster are transmitted and received in a digital format. In the home, such signals are received by a *set-top box* and prepared for presentation upon the television screen. In contrast, any definition of ITV must include the possibility for the end-user to interact with content which is being delivered (or has been delivered) to the home.

For such interaction to be possible, the ”television system” must be able to *receive feedback* from the end-user, as well as process the feedback and/or deliver the feedback upstream for further processing. As a result, today’s television must be augmented with new functionality and features. Like Digital TV, approaches for augmenting television functionality have typically been implemented through the addition of a set-top box. A set-top box can be conceived of as a special purpose PC, since it has specialized channels for input and output, a central processing unit and some kind of memory and/or cache. Like PCs, set-top boxes can be more or less powerful, depending upon their local processing power, functionality and storage capacity. It is certainly conceivable that in the future, the difference between set-top boxes and today’s PCs will nearly vanish. Further, this component-pair may ultimately be embedded within the television itself.

In ITV, *local interaction* can be defined to be limited to the interaction between an end-user and contents and/or applications which have been delivered and stored in the set-top box. Local interaction can be supplemented with *net-based interaction*; in this case, a *return channel* exists in which feedback from the end-user can be sent back to the original information provider (or perhaps some other information/service provider). Typical approaches for implementation of the return channel include modem-based connections to telephone resources and television cable.

18.2 Major issues and challenges

Within the realm of Interactive TV, at least four major areas of challenge can be identified:

- technological challenges
- content- and service-related challenges
- usability challenges and
- challenges in the area of policy-making and business.

Technological challenges in ITV arise from the continued development of conflicting / incompatible standards and hardware platforms; the lack of standardized APIs; the possibilities for alternative transmission channels and data formats; the need for security and privacy mechanisms (e.g., via the use of smartcards); the introduction of new kinds of devices for remote control and input; and, the possibilities for advanced coordination of ITV-based information with other devices in the home.

Some of the **content- and service-related challenges** include: new possibilities for content / program production; information-packaging; electronic program guide (EPG); service access; support for higher-level applications such as learning and surveillance; and, the conception and development of new, innovative services (e.g., "see-and-shop", anytime banking, media-on-demand, virtual libraries, professional consultation, information bureaus, etc.)

Some **challenges in usability** are: *environmental factors* (e.g., for use in the living room, visibility/legibility of details from a distance); *user interface* (e.g., new kinds of "browsers", organization of content and services upon the screen); *remote operation* of the television system (via new kinds of wireless input devices); *customization* of user information and layout (via profiles, information filters, etc.); and, *coordination* with other devices in the home.

Policy and business challenges in ITV include determination as to which actors have access to which information within the return channel, and the design of business models.

18.3 Area(s) of focus

Within the realm of ITV challenges, NR has interest in pursuing:

- opportunities in the area of new production forms, as well as architectures for production-side archiving, indexing, retrieval and flexible packaging of multimedia content
- user-near technological issues
- archiving approaches / architectures for the set-top box
- the creation and development of innovative services
- user interface design and interaction models and
- designs for support of higher-level services (e.g., learning, surveillance, remote control within hazardous environments, etc.).

To address areas like these, NR has existing competence — and is developing new knowledge — in the areas of media and transmission formats, broadband networks, quality of service, design of interactive applications and services, digital television, multimedia production, user interface and usability issues and systems design.

REFERENCE MATERIALS

19. "The Case for IPv6" (chapter 1 only)

Internet Architecture Board
INTERNET DRAFT

Steve King, Bay Networks
Ruth Fax, Bay Networks
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The Case for IPv6
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Status of This Memo

This document is a submission by the IAB Working Group of the Internet Engineering Task Force (IETF). Comments should be submitted to the iab@isi.edu mailing list.

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Abstract

This document outlines the business and technical case for IPv6. It is intended to acquaint both the existing IPv4 community with IPv6, to encourage its support for change, and to attract potential future users of Internet technology.

Modified file for IKT project: IPv6-Summary

Main changes are:
Chapter 2 - The technical case for IPv6 - deleted
Chapter 3 - Transition Scenaries - deleted

Contents

Introduction

This document was produced at the request of the IAB, based on an existing original. The base protocol specifications are now Draft Standards, and are thus unlikely to change. Some other related specifications are still in progress at the time of this writing, so that the technical details are subject to change, and the references cited may become obsolete; as with IPv4, there will always be more work to do. The intended audience includes enterprise network administrators and decision makers, router vendors, host vendors, Internet Service Providers (ISPs) managers, and protocol engineers who are as yet unfamiliar with the basic aspects of IPv6.

The Internet Protocol (IP) has its roots in early research networks of the 1970s, but within the past decade has become the leading network-layer protocol. This means that IP is a primary vehicle for a vast array of client/server and peer-to-peer communications, and the current scale of deployment is straining many aspects of its twenty-year old design [4].

The Internet Engineering Task Force (IETF) has produced specifications (see section 1.1) that define the next-generation IP protocol known as "IPng," or "IPv6." IPv6 is both a near-term and long-range concern for network owners and service providers. IPv6 products have already come to market; on the other hand, IPv6 development work will likely continue well into the next decade. Though it is based on much-needed enhancements to IPv4 standards, IPv6 should be viewed as a new protocol that will provide a firmer base for the continued growth of today's internetworks.

Because it is intended to replace IP (hereafter called IPv4) IPv6 is of considerable importance to businesses, consumers, and network access providers of all sizes. IPv6 is designed to improve upon IPv4's scalability, security, ease-of-configuration, and network management; these issues are central to the competitiveness and performance of all types of network-dependent businesses. IPv4 can be modified to perform some of these functions, but the expectation within the IAB is that the results are likely to be far less useful than what could be obtained by widespread deployment of IPv6. On the other hand IPv6 aims to preserve existing investment as much as possible. End users, industry executives, network administrators, protocol engineers, and many others will benefit from understanding the ways that IPv6 will affect future internetworking and distributed computing applications.

By early 1998 a worldwide IPv6 testing and pre-production deployment network, called the 6BONE, had already reached approximately 400 sites and networks in 40 countries. There are over 50 IPv6 implementations completed or underway worldwide, and over 25 in test or production use on the 6BONE. The 6BONE has been built by an active population of protocol inventors, designers and programmers. They have worked together to solve the questions and problems that might be expected to arise during such a huge project. Their experience has served to validate the expectations of the protocol designers.

This document presents IPv6 issues in several parts:

- The Business Case for IPv6, giving a highlevel view of business issues, protocol basics, and current status, and
- The Technical Case for IPv6, which describes more of the functional and technical aspects of IPv6.
- Transition Scenarios, which discusses mechanisms that have been designed to ease the transition from IPv4 to IPv6.

1. The Business Case for IPv6

Given the remarkable growth of the Internet, and business opportunity represented by the Internet, IPv6 is of major interest to business interests, enterprise internetworks, and the global Internet. IPv6 presents all networking interests with a opportunity for global improvements, which is now receiving the collective action that is needed to realize the benefits.

1.1. IPv6: Standardization and Productization Status

IPv6, the Next-Generation Internet Protocol, has been approved as a Draft Standard, so that it is known to be highly stable and appropriate for productization. A large number of end-user organizations, standards groups, and network vendors have been working together on the specification and testing of early IPv6 implementations. A number of IETF working groups have produced IPv6 specifications that are finished or well underway. Current Draft Standards include:

- RFC 2373: IP Version 6 Addressing Architecture
- RFC 2374: An IPv6 Aggregatable Global Unicast Address Format
- RFC 2460: Internet Protocol, Version 6 (IPv6) Specification
- RFC 2461: Neighbor Discovery for IP Version 6 (IPv6)
- RFC 2462: IPv6 Stateless Address Autoconfiguration
- RFC 2463: Internet Control Message Protocol (ICMPv6) for the Internet Protocol Version 6 (IPv6) Specification

Current Proposed Standards include:

- RFC 1886: DNS Extensions to support IP version 6
- RFC 1887: An Architecture for IPv6 Unicast Address Allocation
- RFC 1981: Path MTU Discovery for IP version 6
- RFC 2023: IP Version 6 over PPP
- RFC 2080: RIPng for IPv6
- RFC 2452: IP Version 6 Management Information Base for the Transmission Control Protocol
- RFC 2454: IP Version 6 Management Information Base for the User Datagram Protocol
- RFC 2464: Transmission of IPv6 Packets over Ethernet Networks
- RFC 2465: Management Information Base for IP Version 6: Textual Conventions and General Group
- RFC 2466: Management Information Base for IP Version 6: ICMPv6 Group
- RFC 2467: Transmission of IPv6 Packets over FDDI Networks
- RFC 2470: Transmission of IPv6 Packets over Token Ring Networks
- RFC 2472: IP Version 6 over PPP
- RFC 2473: Generic Packet Tunneling in IPv6 Specification
- RFC 2507: IP Header Compression
- RFC 2526: Reserved IPv6 Subnet Anycast Addresses
- RFC 2529: Transmission of IPv6 over IPv4 Domains without Explicit Tunnels
- RFC 2545: Use of BGP-4 Multiprotocol Extensions for IPv6 Inter-Domain Routing
- RFC 2590: Transmission of IPv6 Packets over Frame Relay

- RFC 2675: IPv6 Jumbograms
- RFC 2710: Multicast Listener Discovery (MLD) for IPv6
- RFC 2711: IPv6 Router Alert Option

There are too many related RFCs and Internet Drafts to list them all here, but among them are included the following:

- RFC 1888: OSI NSAPs and IPv6
- RFC 2292: Advanced Sockets API for IPv6
- RFC 2375: IPv6 Multicast Address Assignments
- RFC 2450: Proposed TLA and NLA Assignment Rules
- RFC 2471: IPv6 Testing Address Allocation
- RFC 2553: Basic Socket Interface Extensions for IPv6
- OSPF for IPv6
- Mobility Support in IPv6
- DHCP for IP Version 6
- Router Renumbering for IPv6
- Site prefixes in Neighbor Discovery
- Routing of Scoped Addresses in the Internet Protocol Version 6 (IPv6)

Standards work on IPv6 and related components is far enough along that vendors have already committed to a considerable number of development and testing projects. All of the major router vendors have made plans to support IPv6 in their products.

Most or all major vendors have likewise begun the task of delivering IPv6 on desktop machines and servers. Many organizations are working on IPv6 drivers for the popular UNIX BSD and Linux operating environments. Network software vendors have announced a wide range of support for IPv6 in network applications and communication software products. Software is available from Microsoft for Windows-based clients.

1.2. IPv6 Design Goals

IPv6 has been designed to enable high performance, scalable internetworks that should operate as needed for decades. Part of the design process involved correcting the inadequacies of IPv4. IPv6 offers a number of enhanced features, such as a larger address space and improved packet formats. Scalable networking requires careful utilization of human resources as well as network resources; so, a great deal of attention has been given to creating auto configuration protocols for IPv6, minimizing the need for human intervention when assigning IP addresses and relevant network parameters such as link MTU. Other benefits relate to the fresh start that IPv6 gives to those who build and administer networks. For instance, a well-structured, efficient and adaptable routing hierarchy will be possible. The following sections give an overview of the improvements that IPv6 brings to enterprise networking and the global Internet.

1.2.1. Addressing and Routing

IPv6 helps to solve a number of problems that currently exist within and between enterprises. On the global scale, IPv6 will allow Internet backbone designers to create a flexible and expandable global routing hierarchy. The Internet backbone, where major enterprises and Internet Service Provider (ISP) networks come together, depends upon the maintenance of a hierarchical address system, similar to that of the national and international telephone systems. Large central-office phone switches, for instance, only need a three-digit national area code prefix to route a long-distance telephone call toward the correct local exchange. The current

IPv4 system also uses an address hierarchy to sort traffic towards networks attached to the Internet backbone.

Without an address hierarchy, backbone routers would be forced to store route table information on the reachability of every network in the world. Given the current number of IP subnets in the world and the growth of the Internet, it is not feasible to manage route tables and updates for so many routes. With a hierarchy, backbone routers can use IP address prefixes to determine how traffic should be routed through the backbone. In recent years, IPv4 has begun to use a technique called Classless InterDomain Routing (CIDR) [30, 15], which uses bit masks to allocate a variable portion of the 32-bit IPv4 address to a network, subnet, or host. CIDR permits "route aggregation" at various levels of the Internet hierarchy, whereby backbone routers can store a single route table entry that provides reachability to many lower- level networks.

But CIDR does not guarantee an efficient and scalable hierarchy. In order to avoid maintaining a separate entry for each route individually, it is important for routes at lower levels of the routing hierarchy, that naturally have longer prefixes, to be collected together (or "summarized") into fewer and less specific routes at higher levels of the routing hierarchy.

Legacy IPv4 address assignments that originated before CIDR and the current access provider hierarchy often do not facilitate summarization. The lack of uniformity of the current hierarchical system, coupled with the rationing of IPv4 addresses, makes Internet addressing and routing quite complicated. These issues affect highlevel service providers and consequently individual end users in all types of businesses. Furthermore, renumbering IPv4 sites when changing from one ISP to another, to maintain and improve address/route aggregation, is unnecessarily complicated (and thus more expensive) compared to IPv6's ease of site renumbering (see section 1.2.3).

1.2.2. Eliminating Special Cases

Many of the same problems that exist today in the Internet backbone are also being felt at the level of the enterprise and the individual business user. When an enterprise can't summarize its routes effectively, it puts a larger load on the backbone route tables. If an enterprise can't present globally unique addresses to the Internet, it may be forced to deploy private, isolated address space that isn't visible to the Internet.

Users in private address spaces with non-unique addresses typically require gateways, and possibly Network Address Translators (NATs) [31], to manage their connectivity to the outside world. In such situations, some services are simply not available. A NAT is meant to allow an enterprise to have whatever internal address structure it desires, without concern for integrating internal addresses with the global Internet. This is seen as particularly convenient in the existing IPv4 world, with its more cumbersome address space management. The NAT device sits on the border between the enterprise and the Internet, converting private internal addresses to a smaller pool of globally unique addresses that are passed to the backbone and vice versa (see Figure 1).

```
                      |
                      |
Private address space | Unique global addresses
                      |
                      |
```

```
 ---------------        |
/               \    +------+     +-----------+
|  Enterprise   |    |      |     |           |
|               |----| NAT  |-----| Internet  |
|    Network    |    |      |     |           |
\               /    +------+     +-----------+
 ---------------        |
                        |
                        |
                        |
```

Figure 1: Network Address Translator (NAT)

NAT may be appropriate in some organizations, particularly if full connectivity with the outside world is not desired. But for enterprises that require robust interaction with the Internet, NAT devices often get in the way. The NAT technique of substituting address fields in each and every packet that leaves and enters the enterprise is very demanding, and presents a bottleneck between the enterprise and the Internet. A NAT may keep up with address conversion in a small network, but as the enterprise’s Internet access increases, the NAT’s performance must increase in parallel. The bottleneck effect is exacerbated by the difficulty of integrating and synchronizing multiple NAT devices within a single enterprise. Enterprises with NAT are less likely to achieve the reliable highperformance Internet connectivity that is common today with multiple routers attached to an ISP backbone in an arbitrary mesh fashion. Furthermore, use of NAT devices takes away the additional element of reliability afforded by the possibility for asymmetric routing, since NAT devices require control of traffic directions both to and from internally addressed network nodes.

NAT translators also run into trouble when applications embed IP addresses in the packet payload, above the network layer. This is the case for a number of applications, including certain File Transfer Protocol (FTP) programs, Mobile IP, and the Windows Internet Name Service (WINS) registration process of Windows 95 and Windows NT. Unless a NAT parses every packet all the way to the application level, it is likely to fail to translate some embedded addresses, which will lead to application failures. NAT can also break Domain Name Servers, because they work above the network layer. NATs prevent the use of IP-level security between the endpoints of a transaction. Today, NAT devices are helpful in certain limited scenarios for smaller enterprises, but are considered by many to be generally disadvantageous for the long-term health of the Internet. See [16] for a fuller discussion about the effects of NAT use on the Internet.

1.2.3. Minimizing Administrative Workload

A major component of today’s network administration involves the assignment of networking parameters to computers and other network nodes, that are needed before they can begin any sort of network operation. Information such as an IP address, DNS server, default router, and other configuration details have to be installed at each network node. In many cases, this is still done by manual configuration, either by the network administration, or worse yet by the users themselves. Recent efforts to shift this administrative load onto departmental servers have focussed on deployment of the Dynamic Host Configuration Protocol (DHCP) [14, 1], but this comes along with its own administrative difficulties.

IPv4’s limitations also aggravate the occasional need in many organizations to renumber network devices -- i.e., assign new IP addresses to them. When an enterprise changes ISPs, it may have to either renumber all addresses to match the new ISP-assigned prefix, or implement Network Address Translation devices (NATs). Renumbering may be indicated when a corporation undergoes a merger or an acquisition with consequent network consolidation. Since routing prefixes are assigned to reflect the routing topology of the enterprise networks and the number of nodes attached to the particular network links, there are two ways that the choice of routing prefixes can become inconvenient or incorrect:

1. The routing prefix can become too long for the administration to be able to increase the number of nodes that can be attached to the particular link, and

2. The ways that the network links are connected together, or are connected to the outside world, can change.

Either of these occurrence would indicate the need to renumber one or more enterprise networks. It would be quite profitable to be able to renumber enterprise networks without requiring expensive downtime for the networks and or the nodes on the network.

Address shortages and routing hierarchy problems threaten the network operations of larger enterprises, but they also affect small sites -- even the home worker who dials in to the office via the Internet. Smaller networks can be completely dropped from Internet backbone route tables if they do not adapt to the address hierarchy, while larger networks may refuse to renumber and cause a larger routing problem for the backbone providers of the Internet. With today's IPv4 address registries, ISPs with individual dial-in clients cannot allocate IP numbers as freely as they wish. Consequently, many dial-in users must use an address allocated from a pool on a temporary basis. In other cases, small dial-in sites are forced to share a single IP address among multiple end systems.

A unique IP address sets the stage for users to gain direct connectivity to other users on the Internet, as determined by local policy. It also simplifies a wide range of productive interactive applications, of which telecommuting and remote diagnostics are only two examples. Today's hierarchy of limited and poorly allocated IPv4 addresses has already caused problems, and will continue to do so as more and more devices of varying capabilities are added to the Internet.

1.2.4. Security

Encryption, authentication, and data integrity safeguards are needed for enterprise internetworking and virtual private networks (VPNs). For these purposes, IPv6 offers security header extensions.

The IPv6 authentication extension header allows a receiver to determine with a high degree of certainty whether or not a packet originated from the host indicated in its source address. This prevents malicious users from configuring an IP host to impersonate another, to gain access to secure resources. Such source-address masquerading (spoofing) is among the techniques that could be used to obtain valuable financial and corporate data, or could give adversaries of the enterprise control of servers for malicious purposes. Spoofing might fool a server into granting access to valuable data, passwords, or network control utilities. IP spoofing is known to be one of the most common forms of denial-of-service attack; with IPv4 it is typically impossible for a server to determine whether packets are being received from the legitimate end node. Some enterprises have responded by installing firewalls, but these devices introduce a number of new problems, including performance bottlenecks, restrictive network policies, and limited connectivity to the Internet or even between divisions of the same company.

IPv6 uses a standard method to determine the authenticity of packets received at the network layer, ensuring that network products from different vendors can use interoperable authentication services. IPv6 implementations are required to support the MD5 and SHA-1 algorithms for authentication and integrity checking to insure that any two IPv6 nodes can interoperate securely. Since the specification is algorithm-independent, other techniques may be used as well.

Along with packet spoofing, another major hole in Internet security is the widespread deployment of traffic analyzers and network "sniffers" which can surreptitiously eavesdrop on network traffic. These generally helpful diagnostic devices can be misused by those seeking access to credit card and bank account numbers, passwords, trade secrets, and other valuable data. In IPv6 privacy (data confidentiality) is provided by a standard header extension for end-to-end encryption at the network layer. IPv6 encryption headers indicate which encryption keys to use, and carry other handshaking information. IPv4 network-layer extensions for this have been defined and are compatible with those for IPv6, but are not yet in wide use.

Both IPv6 security headers can be used directly between hosts or in conjunction with a specialized security gateway that adds an additional level of security with its own packet signing and encryption methods.

1.2.5. Mobility

IPv4 has difficulties managing mobile computers, for several reasons:

- A mobile computer needs to make use of a forwarding address at each new point of attachment to the Internet, and it's not always so easy to get such an address with IPv4

- Informing any agent in the routing infrastructure about the mobile node's new location requires good authentication facilities which are not commonly deployed in IPv4 nodes.

- In IPv4, it may be difficult for mobile nodes to determine whether or not they are attached to the same network.

- It is unlikely in IPv4 that mobile nodes would be able to inform their communication partners about any change in location.

Each of these problems is solved in a natural way by using features in IPv6. The benefits for mobile computing are apparent in quite a number of aspects of the IPv6 protocol design, and go beyond merely providing dial-up support for road warriors. The improvements in option processing for destination options, autoconfiguration, routing headers, encapsulation, security, and anycast addresses all contribute to the natural design of mobility for IPv6 [20]. In fact, some satellite work in Europe is already starting to become IPv6 based. The IPv6 mobility advantage may be further emphasized by combining flow label management to provide better Quality of Service to mobile nodes.

1.3. The IPv6 solution

IPv6, with its immensely larger address space, defines a multi-level hierarchical global routing architecture. Using CIDR-style prefixes [30], the IPv6 address space can be allocated in a way that facilitates route summarization, and controls expansion of route tables in backbone routers. The vastly greater availability of IPv6 addresses eliminates the need for private address spaces. ISPs will have enough addresses to allocate to smaller businesses and dial-in users that need globally unique addresses to fully exploit the Internet. Using an example from crowded telephone networks, one might say that IPv6 eliminates the need for "extensions", so that all offices have direct communication lines and do not need operators (automatic or otherwise) to redirect calls.

1.3.1. Address Autoconfiguration

Each IPv6 node initially creates a local IPv6 address for itself using "stateless" address autoconfiguration, not requiring a manually configured server. Stateless autoconfiguration further makes it possible for nodes to configure their own globally routable addresses in cooperation with a local IPv6 router. Typically, the node combines its 48 or 64 bit MAC (i.e., layer-2) address, assigned by the equipment manufacturer, with a network prefix it learns from a neighboring router. This keeps end user costs down by not requiring knowledgeable staff to properly configure each workstation before it can be deployed. These costs are currently part of the initial equipment expense for almost all IPv4 computing platforms. With the possibility of low or zero administrative costs, and the possibility of extremely low cost network interfaces, new market possibilities can be created for control of embedded computer systems. This feature will also help when residential networks emerge as an important market segment.

IPv4 networks often employ the Dynamic Host Configuration Protocol (DHCP) to reduce the effort associated with manually assigning addresses to end nodes. DHCP is termed a "stateful" address configuration tool because it maintains static tables that determine which addresses are assigned to newly connected network nodes. A new version of DHCP has been developed for IPv6 to provide similar stateful address assignment as may be desired by many network administrators. DHCPv6 [2, 27] also assists with efficient reconfiguration in addition to initial address configuration, by using multicast from the DHCP server to any desired population of clients.

The robust auto configuration capabilities of IPv6 will benefit internetwork users at many levels. When an enterprise is forced to renumber because of an ISP change, IPv6 auto configuration will allow hosts to be given new prefixes, without even requiring manual reconfiguration of workstations or DHCP clients. This function also assists enterprises in keeping up with dynamic end-user populations. Autoconfiguration allows mobile computers to receive valid forwarding addresses automatically, no matter where they connect to the network.

1.3.2. IPv6 Header Format

IPv6 regularizes and enhances the basic header layout of the IP packet (see Figures 5,6 in section 2.1). In IPv6, some of the IPv4 header information was dropped or made optional. The simplified packet structure is expected to offset the bandwidth cost of the longer IPv6 address fields. The 16-byte (128-bit) IPv6 addresses are four times longer than the 4-byte IPv4 addresses, but as a result of the retooling, the total IPv6 header size is only twice as large; many processing aspects are substantially more efficient. Note that a number of other designs were considered, including variable length addresses; in the end, simplicity won out over infinite extensibility, partially because 128 bits offers such a huge total address space. Recent work [13] in IP header compression promises to reduce or perhaps even effectively eliminate any additional network load associated with the use of 128-bit addresses over low-bandwidth links.

IPv6 encodes IP header options in a way that streamlines the forwarding process. Optional IPv6 header information is conveyed in independent "extension headers" located after the IPv6 header and before the transport-layer header in each packet. Most IPv6 extension headers are not examined or processed by intermediate nodes (in contrast with IPv4). This enables a big improvement in the deployability of optional IPv6 features, compared to IPv4 where IP options typically cause a major performance loss for the packet at every intermediate router. IPv6 header extensions are variable in length and can contain more information than before. Network protocol designers can introduce new header options in a straightforward manner. More details about the comparisons between the IPv4 and IPv6 headers are discussion in section 2.1.

So far, option fields have been specified for carrying explicit routing information created by the source node, as well as for mobility, authentication, encryption, and fragmentation control. At the application level, header extensions are available for specialized end-to-end network applications that require their own header fields within the IP packet.

1.3.3. Multicast

```
                             Multicast Source
                                  +---+
                                  |   |
                                  |   |
                                  +-+-+
                                    |
                                    |
                                    |
      ---+------+----+----------+---+----+-----+--------+------+-----+-
         |      |    |          |        |     |        |      |     |
         |      |    |          |        |     |        |      |     |
         |      |    |          |        |     |        |      |     |
         |    +-+-+  |          |      +-+-+   |        |      |     |
         |    |   |  |          |      |   |   |      +-+-+    |     |
         |    |   |  |          |      |   |   |      |   |    |   +-+-+
       +-+-+  +---+  |        +-+-+    +---+   |      |   |    |   |   |
       |   |         |        |   |            |      +---+    |   |   |
       |   |       +-+-+      |   |          +-+-+  Multicast  |   +---+
       +---+       |   |      +---+          |   |    Group    |
     Multicast     |   |    Multicast        |   |    Member +-+-+
       Group       +---+      Group          +---+           |   |
       Member                 Member                         |   |
                                                             +---+
                                                           Multicast
                                                             Group
                                                             Member
```

Figure 2: Multicast in Action

Modern internetworks need to transmit streams of video, audio, animated graphics, news, financial, or other timely data to groups of functionally related but dispersed endstations. This is best achieved by network layer multicast. Typically, a server sends out a single stream of multimedia or time-sensitive data to be received by subscribers. A multicast-capable network routes the server's packets to each subscriber in the multicast group using an efficient path (see Figure 2), replicating only as needed. In the figure, a single packet from the source will be received by all the multicast group members. When there are multiple networks containing multicast group members, a packet distribution "tree" is created for the multicast group.

Routers use multicast protocols such as DVMRP (Distance Vector Multicast Routing Protocol) [11] and PIM (Protocol Independent Multicast) [9] or MOSPF (Multicast Open Shortest Path First) [24] to dynamically construct the packet distribution tree that connects all members of a group with the multicast server. Only members that have joined the multicast group perform the processing to receive the data. A new member becomes part of a multicast group by sending a "join" message to a nearby router. The distribution tree is then adjusted to include the new route. Servers can then multicast a single packet, and it will be replicated as needed and forwarded through the internetwork to the multicast group. This conserves both server and network resources and, hence, is superior to unicast and broadcast solutions. Multicast applications have been developed for IPv4, but IPv6 extends IP multicasting capabilities by defining a much larger multicast address space. All IPv6 hosts and routers are required to support multicast. In fact, IPv6 has no broadcast address as such; it has various multicast addresses of various scopes. The improved scoping offered in IPv6 promises to simplify the use and administration of multicast in many applications.

1.3.4. Anycast

Anycast services, supported in the IPv6 specification, are not defined architecturally in IPv4. Conceptually, anycast is a cross between unicast and multicast: an arbitrary collection of nodes may be designated as an anycast group [26]. A packet addressed to the group's anycast address is delivered to only one of the nodes in the group, typically the node with the "nearest" interface in the group, according to current routing protocol metrics. This is in contrast with multicast services, which deliver packets to all members of the multicast group. Nodes in an anycast group are specially configured to recognize anycast addresses, which are drawn from the unicast address space [19].

Anycasting is a new service, and its applications have not been fully developed. Using anycast, an enterprise could forward packets to exactly one of the routers on its ISP's backbone (see Figure 3). If all of a provider's routers have the same anycast address, traffic from the enterprise will have several redundant access points to the Internet. And if one of the backbone routers goes down, the next nearest device automatically will receive the traffic.

```
    -----  -----  -----
    | X |  | Y |  | Z |
    -----  -----  -----
       \    |     /                ------- ISP transit domain ---------
        \   |    /                 |                                  |
         -------                   |              -------             |
         | rtr |----------------------------------| rtr |             |
         -------                   |              -------             |
        /       \                  |             /       \            |
       /         \                 |            /         \           |
-------           -------          |     -------           -------    |
| rtr | Enterprise| rtr |----------------| rtr |  Anycast  | rtr |    |
-------  Network  -------          |     -------   Group   -------    |
       \         /                 |            \         /           |
        \       /                  |             \       /            |
         -------                   |              -------             |
         | rtr |----------------------------------| rtr |             |
         -------                   |              -------             |
            |                      |                                  |
          -----                    |                                  |
          | Q |                    ------- ISP transit domain ---------
          -----
```

Figure 3: Anycast Addressing

In figure 3, suppose some hosts Q, X, Y, and Z in an Enterprise Network send data to the anycast address served by the backbone routers in the Anycast Group of the ISP Transit Domain. The border routers in the Enterprise Network forward the data just as they would for data sent to a unicast address. Then, any one of the backbone routers in the Anycast Group may receive the data, eliminating the overhead which would have been incurred if the backbone routers were instead configured to form a multicast group. If there are multiple home agents for mobile nodes on a single home network, they also join a anycast group. In that way, a mobile node can register with exactly one home agent even in the case when it doesn't know the address of any specific one.

Anycast has been proposed to allow endstations to efficiently access well-known services, mirrored databases, Web sites, and message servers. It can provide a versatile and cost-effective model for enabling application robustness and load balancing. For instance, anycast could provide enterprise robustness by assigning all the DNS servers in an enterprise the same anycast address.

1.3.5. Quality of Service

IPv4 carries a "differentiated services" byte and IPv6 carries an equivalent "traffic class" byte, intended for support of simple differentiated services. Both IPv4 and IPv6 can support the RSVP protocol for more complex quality of service implementations. Additionally, the IPv6 packet format contains a new 20-bit traffic-flow identification field that will be of great value to vendors who implement quality-of-service (QoS) network functions. Such QoS products are still in the planning stage, but IPv6 lays the foundation so that a wide range of QoS functions (including bandwidth reservation and delay bounds) may be made available in a open and interoperable manner.

An additional benefit for QoS in IPv6 is that a flow label has been allocated within the IPv6 header that can be used to distinguish traffic flows for optimized routing. Furthermore, the flow label can be used to identify flows even when the payload is encrypted (i.e., the port numbers are hidden).

1.3.6. The Transition to IPv6

The transition from IPv4 to IPv6 could take one of several paths. Some are lobbying for rapid adoption of IPv6 as soon as possible. Others prefer to defer IPv6 deployment until the IPv4 address space is exhausted, or until other issues leave no other choice. Either way, given the millions of existing IPv4 network nodes, IPv4 and IPv6 will coexist for an extended period of time.

Therefore, IETF protocol designers have gone to great lengths to ensure that hosts and routers can be upgraded to IPv6 in a graceful, incremental manner. The transition will prevent isolation of IPv4 nodes, and also prevent "fork-lift" upgrades for entire user populations. Transition mechanisms have been engineered to allow network administrators flexibility in how and when they upgrade hosts and intermediate nodes. IPv6 can be deployed in hosts first, in routers first, or, alternatively, in a limited number of adjacent or remote hosts and routers. The nodes that are upgraded initially do not have to be colocated in the same local area network or campus.

Many upgraded hosts and routers will need to retain downward compatibility with IPv4 devices for an extended time period (possibly years or even indefinitely). It was also assumed that upgraded devices should have the option of retaining their IPv4 addresses. To accomplish these goals, IPv6 transition relies on several special functions that have been specified by the ''ngtrans'' working group of the IETF, including dual-stack hosts, routers, and tunneling IPv6 via IPv4. A dual-stack host is a computer able to handle both IPv4 and IPv6 packets. Such a computer can deliver packetized data to a single application that has been equipped to ask for data from both addressing domains. This facilitates easy transition from IPv4 to IPv6 since the application can then still receive data from its current communications partners, without change in any way noticeable to the users.

1.3.7. IPv6 DNS

Domain Name Service (DNS) is something that administrators must consider before deploying IPv6 or dual-stack hosts. In response to this issue, IETF designers have defined "DNS Extensions to Support IP Version 6" [33]. This specification creates a new "AAAA" (quad A) DNS record type that will map domain names to an IPv6 address. Domain name lookups (reverse lookups) based on 128-bit addresses also are defined. Once an IPv6-capable DNS is in place, dual-stack hosts can interact interchangeably with IPv6 nodes. If a dual-stack host queries DNS and receives back a 32-bit address, IPv4 is used; if a 128-bit address is received, then IPv6 is used. Where the DNS has not been upgraded to IPv6, hosts can resolve name-to-IPv6-address mappings through the use of manually configured local name tables.

IPv6 autoconfiguration and IPv6 DNS can be linked by using dynamic DNS updates, coupled with secure DNS. By these means DNS servers can be securely and automatically updated whenever an IPv6 node acquires a new address, enabling an additional measure of convenience compared with renumbering in IPv4 today.

1.3.8. Application Modification for IPv6

Applications that do not directly access network functions (i.e. do not call a socket or DNS API and do not handle numeric IP addresses in any way) need no modifications to run in the dual-stack environment. Applications that use certain interface APIs to communicate with the network stack will require updating before using IPv6. For example, applications that access DNS or use sockets must be enhanced with the capability to handle AAAA records and 128-bit addresses. Applications which are expected to run both IPv4 and IPv6, as well as using IPv6 security, quality of service, and other features, will need more extensive updating.

Adding such a dual-stack architecture to all the existing hosts is, in fact, a significant effort. This effort has to be balanced against the benefits of IPv6, and against the effort to renumber the existing hosts if the network deployment grows past the restrictions resulting from insufficient address space.

1.3.9. Routing in IPv6/IPv4 Networks

Routers running both IPv6 and IPv4 can be administered in much the same fashion that IPv4-only networks are currently administered. Multi-protocol extensions to BGP4 have been defined by the IETF; one of them carries IPv6 prefixes. The IPv6 extension has been used widely in the 6bone since early 1997. It has been implemented by all the major router vendors and by the well known gated daemon, and is described in a Standard Track document. IPv6 versions of other popular routing protocols, such as Open Shortest Path First (OSPF) and Routing Information Protocol (RIP), are already running.

Administrators may choose to keep the IPv6 topology logically separate from the IPv4 network, even though both run on the same physical infrastructure, allowing the two to be administered separately. Alternatively, it may be advantageous to align the two architectures by using the same domain boundaries, areas, and subnet organization. Both approaches have their advantages. A separate IPv6 architecture can be used to replace the inefficient IPv4 topologies burdening many of today's enterprises. An independent IPv6 architecture presents the opportunity to build a fresh, hierarchical network address plan that will facilitate connection to one or more ISPs. This simplifies renumbering, route aggregation (summarization), and other goals of a routing hierarchy.

Initially, many IPv6 hosts may have direct connectivity to each other only via IPv4 routers. Such hosts will exist in islands of IPv6 topology surrounded by an ocean of IPv4. So, there are transition mechanisms that allow IPv6 hosts to communicate over intervening IPv4 networks. The essential technique of these mechanisms is IPv6 over IPv4 tunneling, which carries IPv6 packets within IPv4 packets (see Figure 4). Tunneling allows early IPv6 implementations to take advantage of existing IPv4 infrastructure without any change to IPv4 components. A dual-stack router or host on the "edge" of the IPv6 topology simply inserts an IPv4 header in front of ("encapsulates") each IPv6 packet and sends it as native IPv4 traffic through existing links. IPv4 routers forward this traffic without knowledge that IPv6 is involved. On the other side of the tunnel, another dual-stack router or host "decapsulates" (removes the extra IP header from) the IPv6 packet and routes it to the ultimate destination using standard IPv6.

To accommodate different administrative needs, IPv6 transition mechanisms include two types of tunneling: automatic and configured. To build configured tunnels, administrators manually define IPv6-to-IPv4 address mappings at tunnel endpoints. Outside of the tunnel, traffic is forwarded with full 128-bit addresses. At the tunnel entry point, a manually configured router table entry dictates which IPv4 address is used to traverse the tunnel. This requires

```
                     +-------------------+
  +-----------+      |   IPv4 Network    |     +-----------+
  | Dual-stack|      |                   |     | Dual-stack|
  | IPv4/IPv6 ========tunnel through======== IPv4/IPv6 |
  | router    |      |                   |     | router    |
  +-----------+      |                   |     +-----------+
     / | \           +-------------------+        / | \
    /  |  \                                      /  |  \
   /   |   \                                    /   |   \
 +--+  +--+  +--+                             +--+  +--+  +--+
 |  |  |  |  |  |                             |  |  |  |  |  |
 +--+  +--+  +--+                             +--+  +--+  +--+
 IPv6 endstations                             IPv6 endstations
```

Figure 4: IPv6 over IPv4 Tunneling

a certain amount of manual administration at the tunnel endpoints, but traffic is routed through the IPv4 topology dynamically, without the knowledge of IPv4 routers. The 128-bit addresses do not have to align with 32-bit addresses in any way.

Mbone deployment using IP-within-IP tunneling has been quite successful, and validates this design approach as well as supporting the likelihood of smooth transition.

1.3.10. The Dual-Stack Transition Method

Initial users of IPv6 machines will require continued interaction with existing IPv4 nodes. This is accomplished with the dual-stack IPv4/IPv6 approach. Many hosts and routers in today's multivendor, multiplatform networking environment already support multiple network stacks. For instance, the majority of routers in enterprise networks are multiprotocol routers. Many workstations run some combination of IPv4, IPX, AppleTalk, NetBIOS, SNA, DECnet, or other protocols. The inclusion of one additional protocol (IPv6) on an endstation or router is a well-understood problem. When running a dual IPv4/IPv6 stack, a host has access to both IPv4 and IPv6 resources. Routers running both protocols can forward traffic for both IPv4 and IPv6 end nodes.

Dual-stack machines can use totally independent IPv4 and IPv6 addresses, or they can be configured with an IPv6 address that is IPv4-compatible. Dual-stack nodes can use conventional IPv4 autoconfiguration services (DHCP) to obtain their IPv4 addresses. IPv6 addresses can be manually configured in the 128-bit local host tables, or preferably obtained via IPv6 autoconfiguration mechanisms. Major servers will run in dual-stack mode until all active nodes are converted to IPv6.

1.3.11. Automatic Tunneling

Automatic tunnels use "IPv4-compatible" addresses, which are hybrid IPv4/IPv6 addresses. A compatible address is created by adding leading zeros to a 32-bit IPv4 address to pad it out to 128 bits. When traffic is forwarded with a compatible address, the device at the tunnel entry point can automatically address encapsulated traffic by simply converting the IPv4-compatible 128-bit address to a 32-bit IPv4 address. On the other side of the tunnel, the IPv4 header is removed to reveal the original IPv6 address. Automatic tunneling allows IPv6 hosts to dynamically exploit IPv4 networks, but it does require the use of IPv4-compatible addresses, which do not bring the benefits of the 128-bit address space.

IPv6 nodes using IPv4-compatible addresses cannot take advantage of the extended address space, but they can exploit the other IPv6 enhancements, including flow labels, authentication, encryption, multicast, and anycast. Once a node is migrated to IPv6 with IPv4 compatibility, the door is open for a fairly painless move to the full IPv6 address space. IPv4-compatible addressing means that administrators can add IPv6 nodes while initially preserving their basic address and subnet architecture. Automatic tunnels are available when needed, but they may not be necessary when major backbone routers are upgraded to include the IPv6 stack. Upgrades can be achieved quickly and efficiently when backbone routers support full remote configuration and upgrade capabilities.

20. “IP and Multi Protocol Label Switching” (extract)

IP and Multi Protocol Label Switching

Extract from Mary Petrosky column in Performance Computing published on Internet PACKETS & PROTOCOLS - 1998/1999

Multiprotocol Label Switching

The growth of the Internet and the rush to increase functionality in IP, the shortcomings of the Internet’s set of protocols and traditional routing methods have become apparent. Ten years ago, it was hard to imagine the Internet and other public networks carrying the volume of traffic they do today. Likewise, the size of today’s networks is a challenge to network designers and operators.

Firstly the Internet service providers (ISPs) feel the pressure. Multinational corporations, state governments, and other large organizations are also finding themselves in the role of service provider, operating large networks to connect their agencies and offices. Like ISPs, these organizations are looking for ways to scale their IP-oriented networks.

Best-effort service and routing via the shortest path (the traditional characteristics of IP networks) are no longer acceptable. If IP-based voice and video services are to become realities, service providers and IT managers need the ability to route timing-sensitive traffic over paths with the lowest latency. These may not be the shortest paths. Similarly, when congestion occurs on a primary network path, operators need a way to shift some of that traffic to a parallel link, or to dump traffic to a preferred alternate route in the case of network failure. None of this is possible with the current IP-routing paradigm.

IETF proposal for MPLS

Fortunately, the Internet Engineering Task Force (IETF) has been working on these problems and the standardization of an architecture and associated protocols for a new IP-forwarding method known as Multiprotocol Label Switching (MPLS).

As of this spring, the IETF had completed work on the core standards for MPLS, including the mechanisms that network operators need to set up explicit paths through a network (vs. hop-by-hop shortest paths). There’s widespread vendor support for MPLS, ranging from heavyweights such as Cisco Systems, Lucent Technologies, and Nortel Networks, to start-up gigabit-router makers such as Juniper Networks Inc. and Torrent Technologies Corp. Products that employ MPLS are just beginning to hit the market, with many more to come in the second half of 1999 and the early part of 2000.

MPLS relies on label swapping or switching for traffic forwarding. Currently, routers and other layer-3 devices forward IP traffic based on matching the IP address in a packet with an entry in the routing table of a layer-3 device. Routers scan the packet header until they arrive at the longest prefix match. In contrast, MPLS allows streams of data to be forwarded based on an exact match with a short, fixed-length label. The labels can also be used to identify classes of traffic, so that streams of data can be forwarded identically. Herein lies one of MPLS’s key strengths.

MPLS Basics

MPLS does not eliminate the need for standard layer-3 routing and traditional routing protocols such as Open Shortest Path First or Border Gateway Protocol. These routing protocols are still needed to determine a packet’s path through a network. However, under MPLS, this route information is bound to a label. Devices operating as MPLS nodes forward packets with the same label in the same way -- they forward it out the same port. MPLS nodes also run special protocols to distribute label information to each other.

A label is a shorthand for a packet header, and is used to index the forwarding decision that a router would make for the packet. However, a label can have various meanings, representing everything from a host route (a full 32-bit IP address) to some combination of source and destination information, or even a Classless Inter-Domain Routing (CIDR) prefix. A label could be used to represent packets that share a common class of service or a particular multicast session, or equate to a given organization’s Virtual Private Network (VPN) traffic.

When a packet first enters an MPLS domain, it’s forwarded using normal layer-3 techniques, with the exception that the outgoing packet includes a label. The job of that first MPLS node includes examining the IP header, classifying the packet, assigning it a label, encapsulating the packet in an MPLS header, and forwarding it to the next hop. At the next hop, the MPLS node performs simple functions: it looks up the label in a table, swaps labels, may decrement a time to live (TTL) field, and forwards the packet.

Each label has only local significance and at each router hop, the router strips off the current label and puts on a new one, forwarding the packet based on the information in the new label. (Asynchronous Transfer Mode [ATM] operates much the same way.) When each MPLS node along a path between a sender and receiver has an appropriate label for each of the next hops in the path, then a label-switched path is established. Under MPLS, a labelswitched path can be based on traditional hop-by-hop routing, or explicitly set up using newly-defined mechanisms for explicit routing.

The advantage of MPLS

As a new type of IP transport, MPLS has the potential to provide operating efficiencies for IP-based traffic, and to support new IP-oriented services. MPLS proponents see it as bringing connection-oriented behavior to traditionally connectionless IP-based networks, opening the door for traffic engineering and related capabilities. For example, with MPLS, service providers could potentially give customers the privacy and quality of service (QoS) of frame relay but with the any-to-any connectivity typical of IP networks.

MPLS promises benefits in four main areas:

- **First,** it can potentially improve the performance of routed networks by providing essentially a new type of layer-2 forwarding. In that it uses shorthand information for packet forwarding, MPLS is a form of short-cut or cut-through routing. Network architects at the Missouri Research and Education Network (MOREnet) are interested in MPLS for precisely this reason.

  By using MPLS its possible to reduce latency in the network by eliminating the "high touch" functions a router performs on each packet. By expediting traffic handling at each network node, the architects hope to run latency-sensitive traffic, such as voice, over their packet network. In addition, MPLS will enable the architects to flatten MOREnet, which is currently a hierarchically routed network. Flattening the network will simplify operations.

- **Second,** MPLS addresses the scalability of today’s large networks. Currently, many service providers and large organizations are growing their networks by deploying a hybrid router-and-ATM switch architecture. In these networks, routers operate at the edges of an ATM switch core. The routers are connected to each other via a mesh of ATM permanent virtual circuits (PVCs).

  Unfortunately, this architecture leads to what’s known as the "n-squared" problem. As networks grow in size, the number of PVCs required grows exponentially, potentially stressing the ATM switch capacity. In addition, each router views all other routers to which it’s connected as peers, and must maintain reachability information for all other routers ringing the ATM core. As a result, router tables can become quite large, and potentially unstable. Likewise, in the event of a route change, convergence times can become lengthy.

  With MPLS, ATM switches can run MPLS control software and operate as peers with routers. Consequently, each MPLS node (whether a router or an ATM switch) needs only be aware of its immediately adjacent neighbors, thus reducing the number of devices with which it is a peer. MPLS also improves network scalability by supporting route aggregation.
- **Third,** MPLS promises new services, including QoS, multicasting, and VPNs. As we noted previously, ISPs could roll out new services simply by changing the way packets are assigned to label switched paths (LSPs). Packets could be assigned to an LSP based on a VPN identifier, a specific QoS requirement, an IP multicast group, or any combination of the destination subnetwork and application type, or a combination of the source and destination subnetworks.

  Today, each router in a packet’s path must analyze the packet’s network-layer header to determine the packet’s precedence or class of service in order to apply the appropriate discard thresholds, queuing, and forwarding policies. MPLS can support QoS by allowing the precedence or class of service to be inferred from the label, so that no further header analysis is needed.

- **Fourth**, MPLS's can as most important for service providers, support traffic engineering. Traffic engineering is essentially the process of selecting paths for data traffic with the goal of balancing the network load on various links, routers, and switches. Traffic engineering gives network operators the ability to route primary paths around known bottlenecks or points of congestion, and to provide more-efficient use of available bandwidth.

  MPLS handles traffic engineering through its use of explicit routing. With explicit routing, an edge node determines the path to the destination and sets up an explicit route, indicating which nodes are on that route. Explicit routes can be based on administrative policies, enabling network operators to select routes with an eye to traffic management, including the loading of the bandwidth through the nodes and links in the network.

  In some cases, network operators can forward certain classes of traffic along certain pre-specified paths, rather than have the traffic follow the hop-by-hop path. For example, a class of service or QoS level can be factored into the establishment of an explicit route, ensuring that traffic receives preferential treatment over best-effort traffic. Likewise, network operators have the ability to aggregate streams of data, which may represent multiple flows of user data, and forward that data as a unit over a single specified path.

The IETF has made significant progress in defining standards for MPLS, however additional related standards are necessary for customers to fully realize MPLS's. The MPLS working group has defined two standards for explicit routing, and vendors have split into camps favoring one approach over the other, raising interoperability questions.

Status of MPLS

Multiprotocol Label Switching (MPLS) holds great promise as a new technology for IP traffic forwarding. This technology brings greater operating efficiencies and new capabilities to IP networks. However, MPLS is merely a budding technology right now. It and many of the technologies it supports need time to mature before MPLS blooms.

MPLS's use of labels can streamline the packet-forwarding process. MPLS labels also can carry class- or quality-of-service (CoS/QoS) information, Virtual Private Network (VPN) information, or other characteristics about the traffic being forwarded. Because this information is in the labels at the front of each packet, layer-3 switches and routers don't need to read into packets to find it, saving time and processing cycles over the current longest-prefix-match forwarding method.

MPLS gives IP networks the ability to set up connections, which opens the door to traffic engineering and explicit routing. These capabilities will give operators of IP-based networks more control over how traffic is routed (and rerouted, in the case of failures). MPLS also will let IP networks select routes that support an application's need for specific class or quality of service. With explicit routing, for example, latency-sensitive traffic, such as voice and video, can be routed over a path that has been engineered for low latency.

During the first half of 1999, the IETF MPLS working group defined many of the core standards for MPLS. In addition to approving the MPLS architecture document, the working group defined the label format and how this information can be carried over Asynchronous Transfer Mode (ATM), frame relay, and 802.3 LAN links. The working group also standardized the way labels are distributed between routers, ATM switches, and other devices acting as MPLS nodes.

Possible Ruting Protocol Contention

The MPLS working group is standardizing more than one approach to explicit routing, which underpins traffic engineering. Although variety is generally a good thing, having multiple solutions to the same technical problem usually leads to interoperability which is likely to be the case with MPLS explicit routing.

In general, MPLS supporters agree there is a need for a basic approach to label distribution. This has been standardized in the form of the Label Distribution Protocol (LDP), which permits hop-by-hop routing. However, the working group is moving ahead with two approaches to explicit routing, each of which has its devotees. One approach, dubbed "constraint-based label switched path setup using LDP" (CR-LDP), is based on extensions to LDP. The other approach is based on a modified version of the Resource Reservation Protocol (RSVP), which was originally defined to support QoS capabilities, such as bandwidth reservation, in an IP environment. Nortel Networks is the key backer of CR-LDP, while Cisco Systems is backing the RSVP approach. Each camp has attracted a few followers, while other players are remaining neutral (meaning they'll implement both).

Both explicit routing schemes allow for explicit route setup, tear down, and re-routing in case of a failure. The two differ in how they operate, however, and proponents for both sides argue their scheme's relative merits. CR-LDPers maintain theirs is a more "friendly" approach for ATM environments, while RSVPers argue that leveraging an existing protocol (like RSVP) will get products to market quicker. Regardless of their pros and cons, the fact remains that there are two approaches. And it's unclear when (or even if) the two will interoperate. For some organizations, this lack of interoperability may be a showstopper. For other organizations, the split over explicit routing won't derail their MPLS deployment plans. Rather, it will simply mean sticking with a single-vendor solution for the time being.

MPLS: Necessary But Not Sufficient

While MPLS offers a means to support traffic engineering and new services, such as CoS/QoS, VPNs, and multicasting, the MPLS protocols alone aren't sufficient to deliver these capabilities. Vendors must fill in a few of these blanks, while the IETF defines protocols in some areas as well. For example, traffic engineering relies on paths that are explicitly set up based on some predefined policy, rather than paths that are set up hop-by-hop across the shortest route between a source and destination. Although MPLS controls explicit routing via specific protocols, the mechanisms, processes, and algorithms used to compute explicitly routed paths are beyond the scope of the MPLS specifications. Consequently, vendors will implement their own mechanisms, with varying levels of success.

On the standards side, the IETF must extend routing protocols, such as Open Shortest Path First and Border Gateway Protocol, to support traffic engineering. For example, routing protocols need to be able to advertise the total bandwidth a given link has, what bandwidth is currently in use, and what bandwidth reservations have been made. The IETF is working on these extensions, but more work remains.

MPLS itself doesn't solve the QoS routing problem. As with traffic engineering, routing protocols must be enhanced so that data is routed in accordance with its CoS/QoS requirements. For example, routing protocols need to know which paths have low latency, so video and voice traffic can be forwarded over these paths, rather than paths with high latency. Routing protocols also need to be aware of congestion and other network activity that can impact CoS/QoS. Finally, the MPLS working group must agree on how to carry CoS/QoS information within the label. Again, the IETF is working on these areas, and again, more work is needed.

As for VPNs, the MPLS working group has seen many proposals regarding VPN support in the MPLS environment. So far, the working group has not taken up any of these proposals, and they may be pushed off to an as-yet-to-be-established VPN working group. As a result, vendors will likely use nonstandard methods to support VPNs in an MPLS environment, at least for the next 12 months.

introduction of MPLS as a New Technology

MPLS needs to go through a maturation process and it will take a year or more for MPLS-specific protocols to mature and stabilize, and for vendors to conclude interoperability testing. In addition, MPLS protocols need to interact with existing layer-3 protocols as well as emerging protocols in the areas of CoS/QoS, VPNs, and multicasting. It will take time for the industry to understand and tweak these interactions.

The industry also needs to become more familiar with how to operate and manage an MPLS network. How does MPLS affect network dynamics? What happens in the interval between when a route changes and a new label is assigned to that route? These questions can only be answered after MPLS has been deployed in large production networks, which should begin in the second half of 1999.

Service providers such as AT&T and UUNet are among MPLS's early adopters and have been actively testing the technology. Only after such testing is concluded, and MPLS is deployed in real-world networks, can we know if it lives up to its claims of scalability and robustness, and how well its traffic engineering and other features work.

Beginning in late 1999, AT&T and other MPLS early adopters will begin offering MPLS-based public network services. These deployments will help the industry begin building the knowledge base it needs to move MPLS out of the early-adopter phase. That process will take through 2000 and into 2001.

Despite the word "multiprotocol" in its name, MPLS currently works with IP traffic only. As a result, MPLS has the most to offer enterprises and service providers that run a preponderance of IP traffic over their WAN backbones. As with any technology, potential MPLS users should evaluate it with an eye to their existing network infrastructures, and the applications they support.

MPLS proponents maintain that it provides a better way to support IP over ATM and frame-relay networks than current methods do. For many enterprises and service providers, this aspect of MPLS is one of its key attractions. For example, vendors will give network operators three approaches to choose from when deploying MPLS over an ATM network

- run MPLS as a separate control plane, operating side-by-side with the ATM control plane (including ATM signaling and other protocols);
- run MPLS and ATM control planes in an integrated fashion (where they share routing information, for example);
- or do away with the ATM control plane altogether, and simply run MPLS software on ATM switches.

The first two options are well-suited to enterprises and service providers that want to exploit MPLS for their IP traffic, but continue to use their ATM infrastructure for voice and legacy traffic. The third option may appeal to network operators who have an ATM infrastructure they want to continue to leverage, but are rapidly moving to IP-only.

Service providers, especially new operators that have the luxury of building their networks from scratch, will find a compelling combination in MPLS and the new class of high-speed, high-port-density gigabit and terabit routers coming to market. MPLS will particularly appeal to service providers that want to offer value-added services, such as differentiated CoS/QoS and VPNs.

However, MPLS has little to offer enterprises and service providers that simply need to move bulk IP traffic, and are satisfied with IP's current best-effort delivery scheme. MPLS also has less to offer enterprises and service providers for whom IP is merely one of several types of traffic on their networks. MPLS can't handle legacy protocols, such as SNA or X.25, or traditional circuit-based traffic, unless this traffic can be encapsulated inside IP.

Although many enterprises are moving to IP as their strategic networking protocol, that process will take time. As one IT manager noted, old protocols such as X.25, RS-232, and -422 are still alive and running over his WAN. Although he's interested in MPLS for his IP traffic, he expects to continue using his ATM backbone to support his legacy traffic.

MPLS has clearly caught the attention of service providers. Enterprises that are not interested in deploying MPLS themselves should be aware of the technology, if only for the services it may enable in the public network arena. It's too early to tell whether MPLS will become the predominant method of IP traffic forwarding in the long run. However, it will bring new capabilities to IP-centric networks once it's had time to mature.

21. The DVP Project

21.1 Project Details

ACTS project AC089

Start Date: 1995-09-01

End Date: 1998-02-28

Duration: 30 months

Project Status: Completed

Project Cost: 9.96 million ECU

Project Funding: 3.48 million ECU

21.2 Project Consortium

The project participants are listed below:

Coordinating Partner

- **GMD - German National Research Center for Information Technology** (D)

Full Partners

- **IRT - Institut für Rundfunktechnik** (D)
- **Horz&Schnepf** (D)
- **RAI Research Center** (I)
- **INTECS Sistemi** (I)
- **INTRACOM S.A.** (GR)
- **Mega Channel** (GR)
- **AMEC - advanced media engineering and consulting** (A)
- **CUI (University of Geneva)** (CH)
- **GRAME** (F)
- **SUMS (University of Toronto)** (CAN)
- **NCSA (University of Illinois)** (USA)
- **Caterpillar** (USA, B)

Associated Partners

- **Deutsche Welle** (D)
- **NDR - Norddeutscher Rundfunk** (D)
- **HR - Hessischer Rundfunk** (D)
- **Papazoglou Video Ltd.** (GR)

Sponsoring Partners

- pixelMotion (CAN)
- Tektronix (D)
- Hewlett Packard (D)

- SGI (D)
- BBC (GB)
- Sony (JP)
- TSR (Television Suisse Roman) (Swiss)

21.3 Project Description

21.3.1 Main Objectives

The DVP project (Distributed Video Production) is an ACTS project (AC089) that aims to take existing broadband technology and develop services (applications) that directly address the needs of the television industry. Several broadcasters are included in the consortium and have committed to providing user input and participating in field trials. Thus DVP has a strong element of user participation.

Distributed video production refers to situations where the cameras, recorders, switches, mixers and other equipment used in video production (or post-production) are located at several sites linked by high bandwidth network connections (see Figure 1). The DVP project will investigate user requirements for several forms of distributed video production and will run a series of trials of a distributed virtual studio, a distributed rehearsal system and a distributed video editing and retrieval system.

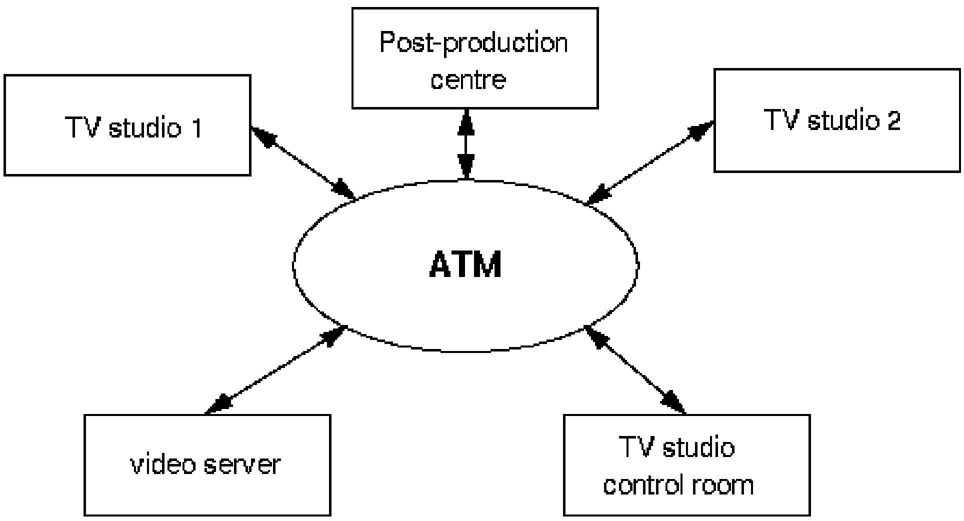

Figure 1 Distributed Video Production

21.3.2 Technical Approach

Basically the DVP project consists of two layered development and field test scenarios (see Figure 2):

- a wide area ATM testbed (including trans-Atlantic links) connected to digital video pro- duction test sites with basic user functions as prerequisite for broadband distribution of digital audio/video applications
- distributed video production pilot applications (e.g. a distributed virtual studio or a dis- tributed video editing or simulation system) as the final application goal.

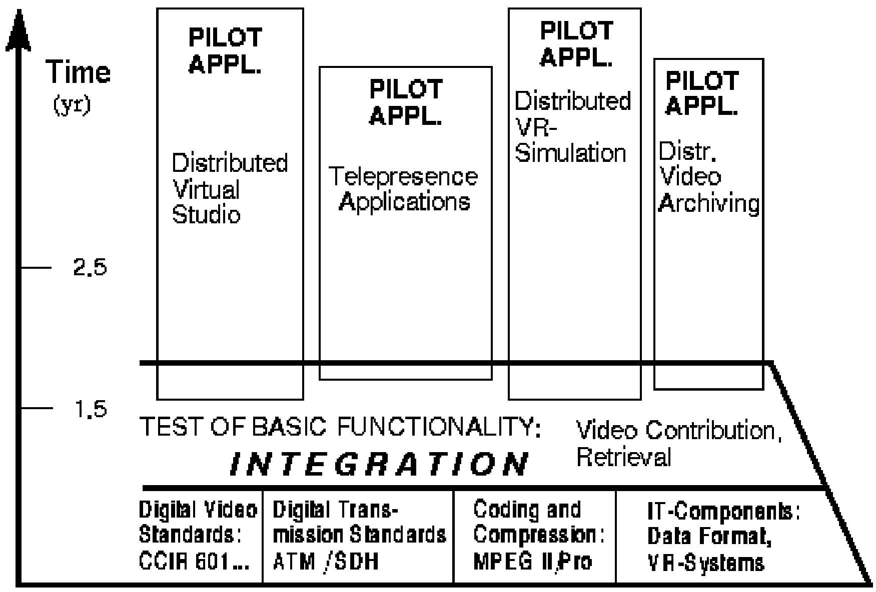

Figure 2 DVP: Project Activities

21.3.3 Key Issues

MPEG II compression has been designed so far mainly for program distribution at relatively low data rates (8 mbs) per stream. There will be a new MPEG II variant for professional use which can be better suited for the video production chain (contribution, production, post-production and distribution). But there is still an open question how compression will influence the video quality if compressed video is used and concatenated in the different production stages.

On the other hand, the usage of compression offers the choice to get video transferred online to places, which could not be interconnected before. New transport layers like ATM imply that the video signal will no more be carried in continuous streams, but will be broken into pieces and packetized into cells. The characteristics of the underlying transfer system are quite different than with isochronous D1 streams, quality of service aspects like cell loss and jitter will be important. But packetization offers the chance that video distribution is no longer limited to local distribution.

In detail, questions like the following ones have to be answered:

- How can ATM-connectivity based on SDH (STM1 to STM4) be used for connecting CCIR 601 studio systems?
- Transmission and processing delays: what is the impact of such delays during video pro- duction, what are acceptable delays, where is isochronous transmission really required?
- Synchronization: what are the synchronization requirements for tightly-coupled dis-tributed video processing?
- What quality of service of the transmission system is required?

21.3.4 Relationship to Previous Work

The DVP project will make use of existing technology in the areas of video encoding, storage and transport systems. It will also build on existing work in such areas such as non-linear ed- iting, video servers, and virtual studios.

21.3.5 Expected Impact

Partners from broadcasting industry participating in a European network would profit from access to DVP facilities for a variety of reasons:

- reduced production costs and shorter production times
- integrated training and documentation facilities
- better (tested/simulated) product quality
- integrated production and distribution processes

In summary, DVP will build advanced communications services that fit the needs of a particular user group. These services will be used in field trials and will contribute to European economic development and social cohesion.

21.3.6 Relationship to Other Projects and Actions

The DVP consortium is in contact with representatives of several related ACTS projects, these are: PANORAMA, SICMA, MIRAGE, AURORA and RESOLV. Areas of mutual interest occur with other European projects, such as the ESPRIT Euromedia project, and with American activities such as the SGI/Sprint Drums service.

21.3.7 Trial Dimension

The project shall identify requirements and develop prototype applications for well defined realistic scenarios. The scenarios comprise a rich set of real life situations, a series of trials will allow end users to judge the potentials of the services provided.

Integrated systems and field trials are an important element of DVP. The prototypes developed in the project are not simply for lab use but will be used under actual working conditions.

Expected trial scenarios are:

- Distributed Video Post-Production: Editing and special effects generation using both local equipment (editing console, DVEs) and remote equipment (video servers).
- Distributed Video Production: Bringing together real actors and objects (props) from different - separated - studios in a common real or virtual studio.
- Distributed Rehearsal: An immersive teleconferencing environment allowing small groups of actors and musicians at different studios to conduct rehearsals as if face-to-face.
- Distributed Video Archiving, Indexing, and Retrieval: Besides real-time applications like those mentioned above, some non-real-time applications will also contribute to DVP.

21.3.8 DVP Public Deliverables

- **1.1:** User Requirements - TV domain. User requirements on distributed video production concerning the TV domain.
- **2.1.1**: State of the Art: Standards and Compression. This report describes current video transmission and encoding technique. Encoding algorithms are explained, corresponding products and the results of subjective quality tests are presented and the relevance for DVP is discussed.
- **2.1.2**: State of the Art: Digital Video and ATM. This report summarizes the current state-of-the-art of video transmission over ATM networks. Several encapsulation schemes and existing standards for video streams are described and evaluated. An overview of ATM products is given.
- **2.2.1**: State of the Art: editing suites. This deliverable provides a survey and an evaluation of existing compressed/uncompressed editing suites.
- **2.2.2**: State of the Art: distributed multimedia architectures. Evaluation of distributed multimedia architectures.
- **2.2.3**: State of the Art: Compression Influence. This report presents the results of a systematic investigation of the influence of compression in the production chain by implementation of many possible chains and subjective tests for selected test sequences.
- **4.1.3**: Pre-trial implementation of a DVS system. This report documents a first implementation of the Distributed Virtual Studio (DVS) Application.
- **4.2.7**: Public report about the DVER system and its evaluation. This report provides a description of the final DVER system, its implementation and a summary of the system evaluation.
- **4.4.3**: Pre-trial implementation of a DVR system. This report documents a first implementation of the Distributed Virtual Reality (DVR) Application.
- **4.4.5**: Public report on the DVR system and its evaluation. This report describes the final DVR system, implementation, measurements and evaluation.
- **5.1**: Distributed Virtual Studio - Trials and Evaluation. This document reports the Distributed Virtual Studio (DVS) trials and gives an evaluation from the technical as well as the economical point of view.
- **6.2.2**: DVP standardization report.
- **6.1.7**: DVP final report

22. High quality Audio over ATM – Report from the Lawo ATM Symposium –99

High quality Audio over ATM – Report from the Lawo ATM Symposium –99

The first non-formal meeting for standardization of professional audio over ATM, Germany October 14–15

Jon Vedum, Telecast Norge a/s
November 27th, 1999

Abstract

Only two standards do currently exist for transfer of professional, linear coded audio. One for two-channel audio, and one for 56 channels. Both are point-to-point connections.

There are therefore a demand for standards for transferring audio over more general networks. The ATM network seems promising because defined Quality of Service is a part of the protocol, and the inherent delay is low. Streaming audio is very critical regarding delay, and especially variations in delay.

One company, Lawo, has already started manufacturing equipment based on ATM. But because they know a standard is the basis for success, they invited a number of competitors to a symposium for discussing a possible standard for audio over ATM.

The participants agreed on a basic format for transferring two-channel audio in a virtual connection. But a lot of details still remain. AES has also started some activities in this field.

22.1 Background

22.1.1 Digital audio in professional applications

Today, compressed digital audio has gained popularity in many applications. A lot of work is being done regarding transportation of compressed audio over general data networks, both as audio files and as bit streams in real time. Often, compressed audio is also handled as a part of a digital video signal. EBU and SMPTE have delivered a comprehensive report regarding exchange of programme material [19], with focus on video.

Compressed audio is of course attractive, especially for transmission over long distances and computer storage. There are, however, at least three disadvantages for using compressed audio in high quality studio applications:

- Cannot, in most cases, directly be mixed or manipulated.
- Almost all compression algorithms introduces delay.
- Almost all compression algorithms introduces artifacts which may be audible in certain circumstances.

For studio and other high-quality applications, linearly encoded audio therefore will be used also in the future. Two standards is currently in widespread use for transport and interfacing linearly represented audio, AES 3 for two-channel and AES 10 for multichannel audio. Additionally, a number of proprietary formats are used.

22.1.2 The AES 3 two-channel Audio interface

Technical description

This is also often mentioned the AES/EBU interface, and is standardised by AES [20], Ansi [21] and EBU [22]. A quite similar interface for consumer applications is standardised by IEC [23].

Some key data for the specification are:

- Audio data format is linearly encoded PCM, 16 – 24 bit. LSB transmitted first.
- Each (mono) sample is represented by 32 bits in any case: 4 bits pre-amble, 24 bits audio, 1 user data bit, 2 status bits and 1 parity bit.
- One *frame* consists of 2 samples. 192 frames makes up a *block*.
- Nominal sampling rate is 48 kHz. 32 kHz (broadcasting) and 44.1 kHz (CD-format) are also accepted for certain applications. 96 kHz is used for some high-quality applications. ±12,5% variation for vari-speed devices may be allowed.
- Physical format is 2-wire balanced 110 ohm cable. Levels almost identical to RS-422, but isolation transformers are specified. Channel encoding scheme, except for the pre-amble, is bi-phase mark.

Problems and limitations

The AES 3 interface is well accepted as a point-to-point connection for a stereo signal. In some applications, i.e. 5.1 surround-systems, more than one cable is needed.

Another limitation is that it is a point-to-point connection. More than one receiver tends to make problems because of loading and reflections.

22.1.3 The AES 10 Multichannel Audio Interface (MADI)

Technical description

The MADI interface, defined in the AES 10 standard [24], is a serial interface format for point-to-point connection of 56 audio channels. The main characteristics are:

- The format for each channel, and also the block format, are similar to the AES 3 format (except for the pre-amble).
- The internal bit frequency is 100 MHz, regardless of audio sampling frequency. After channel encoding, the frequency of the bit stream is 125 MHz.
- The channel encoding scheme is 4/5 bit NRZI, and the physical medium is 75 ohm coax or optical fiber. The interface is based on the FDDI standard.
- For synchronization of audio data, a separate Digital Audio Reference Signal (DARS) is normally needed. (The DARS normally is an empty AES 3 data stream).

The somewhat strange number of channels, 56, is given by:

56 * 32 bit * 48 kHz * 112,5% = 96.768 MHz.

In addition, some synchronisation words are added. This allows for 12,5% overspeed. In the original application of the interface, as an interconnect between a multi-cannel tape recorder and a mixing console, this made sense. However, if only 48 kHz sampling is used the number of channels can be increased to 64. By reducing the sampling frequency to 44.1 kHz, 72 channels can be squeezed through the interface.

Problems and limitations

The interface has never got the same wide-spread acceptance as the AES 3. The reasons probably are both that the demand for a multi-channel connection is less than for a stereo, and a number of technical limitations:

- The number of channels is fixed, no flexibility regarding physical data rate. MADI is therefore not cost-effective for small number of channels.
- A separate synchronization signal is needed. In practical applications, this often means a separate cable along the fiber or coax.
- MADI is a fixed point-to-point connection. No good possibilities for networking.
- The original design was based on the AMD TAXI-chip. This chip is now obsolete.

22.2 The next generation audio interface

22.2.1 Demand for a networking solution

Both AES 3 and MADI are point-to-point connections, based on standards used in the audio world only. More and more people want a common infrastructure both for audio, video and data. This is especially true for wide area applications.

But also for local installations, use of standard data cabling and other equipment is desirable. More standardised infrastructure means simpler installation and maintenance.

Another demand is routing of metadata, i.e. remote control of equipment, together with the audio. There is, of course, a limited number of user bits available in the AES 3 bit stream. But no protocol is defined, and the data rate quite limited.

22.2.2 Real-time audio over general-purpose data networks

Most networks are designed for file transfer

Most of the data networks in use today are optimised for file transfer and other burst applications. They are based on packet switching or frame relay technology, and works well for transfer of relatively large data packets. This is also a good solution for exchange of audio files, but not for real-time streaming audio.

Basically, most networks are also based on “best effort”- algorithms. Packets are delivered as fast as possible, but no guarantee is given for the delay. In many cases, packets may also be thrown away because of network congestion etc.

Basic requirements for streaming audio

The most important technical requirements for transfer of real-time audio are:

- Defined, and high, quality of service. Necessary bandwidth must be available all the time, and almost no variation in delay is accepted. Even short breaks in the transfer are very annoying, loss of a single sample may be audible.
- Low absolute delay. Especially in recording or broadcasting of live events, almost no delay is acceptable. For local connections, the practical limit is found to be in the range 2-5 ms. This also implies that the error rate of the connection must be very low, there is no time for retransmission of corrupted samples.

Experience shows that high quality audio is more demanding than video [25].

Another requirement is of course reasonable cost.

22.2.3 ATM – The ultimate solution?

Technical qualities

Among the many networking technologies available, the Asychronous Transfer Mode technology has some qualities making it a promising alternative for audio:

- Can deliver defined quality of service.
- Low and constant delay because of small data packets (cells) and connection-oriented data transfer.
- Scalable speed, 25 Mb/s to Gb/s.
- Bandwidth on demand. Different bandwidths can be mixed in one network.
- Standard equipment available. Same technology both for local point-to-point connections, and as backbone in world-wide telecom networks.

The Lawo application

One of our major competitors, LAWO AG in Germany, has realised an interesting digital audio mixer based on ATM technology, the mc^2 family. This mixer is a part of what they call DSN, Distributed Studio Network [26].

The heart of the Lawo mixer is an IBM router frame, equipped with both standard ATM cards and specialised interface- and DSP-cards designed by Lawo.

No success without a standard

Currently, no standard for transfer of linearly coded audio over ATM do exist. Lawo has made some preliminary decisions in order to design their products. But they have realised that unless a industry standard is developed, the DSN will probably be no success. They therefore invited a number of competitors, and also some consultants, to an ATM Symposium at their premises in October this year.

22.3 The Lawo ATM Symposium

22.3.1 The purpose of the symposium

As Mr. Philipp Lawo said in his opening speech, they wanted to take an initiative to start the standardisation process for application of ATM in the audio industry. They were willing to share their experiences and current interface standard with everyone, but not of course their implementation. They did not show any equipment during the symposium.

22.3.2 The participants

Invitations was sent by Lawo to a rather random selection of pro-audio manufacturers. Others, like Telecast, contacted Lawo and got an invitation on request. Some consultants with experience in ATM and audio was also invited. In total, approx. 20 people participated.

All participants were skilled engineers in audio technology, but their experience and knowledge about ATM were more diverse.

22.3.3 The agenda

The duration of the symposium was two short days. The first day was an introduction to

ATM, and the Lawo concept. The second day was used for discussions.

Much of the first day is already presented above. I can see no reason for a separate presentation of the Lawo interface, the interesting part is included in the discussion below.

22.3.4 The scope of a standard – Possible applications

All participants agreed to concentrate on real-time streaming audio, and not to discuss transfer of audio-files.

The following main application areas were discussed:

- Wide Area Networking, audio transfer over the public network.
- Local Area Networking, audio transfer within buildings and/or campus.
- Local routing, i.e. between a number of stage-boxes and a mixer, or a number of mixers.
- MADI replacement, point-to-point connection.

22.3.5 The audio format

The format of an audio sample

In some applications, a very high dynamic range is desirable. The most efficient way of transferring audio will then be as floating-point numbers. But today floating-point formats are normally used only inside processing equipment, and no common standard for exchange of audio samples do exist. The group therefore agreed to stick to the current practice:

Resolution: *The format of each audio sample shall be fixed-point binary 2's complement non-compressed linear PCM, with a word length of up to 24 bits.*

How many audio channels in a virtual connection?

One of the basic requirements for a new standard is flexibility regarding the number of audio channels over one physical connection. This gives two possibilities:

1. Few audio channels in each virtual connection, use a number of virtual connections in parallel for larger number of channels.
2. Variable number of channels multiplexed into a virtual connection.

Alternative 2) can be most efficient, but will probably be complex both to describe and implement. The group therefore decided to go for alternative 1), but with option for adding alternative 2) in the future.

What number is "few"? The ultimate flexibility is given by packing each channel in a virtual connection. In order to get efficient use of bandwidth, then at least 12 samples (32 bit * 12 = 48 byte) is to be packed into one ATM cell. This involves a buffer delay of minimum 12/48000 Sec. = 250 µSec., which is more than desired in some applications. This delay is halved if doubling the number of channels. Two channels in one connection will also make better compatibility to the AES 3 interface.

Resolution: *Each virtual connection shall contain two audio channels. Other numbers may be allowed in the future.*

Basic frame format.

In addition to audio, some user data and status information may be necessary. This is a very complex field with a lot of possibilities. In order not to get lost in an eternal discussion, the group agreed to stick to the basic frame format defined in AES 3. This will ensure transparent transfer of AES 3 bit streams. All details regarding use of pre-amble was not specified in the meeting.

Resolution: *The basic frame format shall be compatible with the AES 3 frame. Each audio channel subframe will consist of 24 bits for audio plus 8 bits for framing, status and user data.*

Quality and reliability – Error detection and correction?

A number of different error types can occur in an ATM network. Most types of errors in an audio stream can only be detected, not corrected. By implementing a forward error correction algorithm, most single-bit errors can be corrected. The price is, however, some overhead and reduced efficiency.

Lawo had made some non-formal tests on multi-node connection in the public ATM network within Germany. In all tests, the bit error rate was very low, $< 10^{-12}$. If this is typical, probably no error correction is necessary. But more data is needed.

Resolution: *Further investigation necessary. Probably error correction shall be an option in WAN application. In local area applications, only simple error detection is sufficient.*

The AAL layer – How to pack samples into cells?

Constant bitrate traffic (CBR) normally uses AAL type 1. The first octet in the AAL 1 payload data unit (PDU, the 48 octets available for user data) is used as a sequence counter, therefore only 47 octets are available for user data. Each audio frame occupies 8 octets (32 bit * 2/8). The number of audio frames in each cell is therefore limited to 5, instead of 6 if all 48 octets had been available.

An alternative is to use AAL 0, i.e. no adaption layer at all. Then all 48 octets are free to audio data. The disadvantage is loss of sequence numbers, but the group agreed that this is not important. The ATM network itself is transparent to the AAL layer, so WAN transfer is no problem.

Resolution: *AAL 0 is used for audio, with 6 AES 3 frames (2 sub-frames) in each cell.*

Phase alignment.

In a multi-channel system, all samples must be in phase. Different virtual connections may be routed differently through the network, and therefore arrive at different times. One simple way to ensure phase coherence is to mark some samples or cells, and then buffer all other samples until they all are in phase.

One proposal is to use one bit in the PT (Payload Type) field in the cell header, and mark each 8th cell. This corresponds to marking each 48. sample, or 1 ms between marks at 48 kHz sampling rate. Phase coherency will then be ensured if difference in delay is less than 0.5 ms.

Resolution: *Marking each 8th cell seems to be a reasonable solution for handling short delays. But a solution for handling delay differences >0.5 ms is needed.*

22.3.6 Signaling and remote control functions

At least a simple control protocol for control of pre-amplifiers, converters and other equipment is needed. Time did not allow further discussions within the symposium.

22.3.7 Physical format

155 Mb/s optical, according to ITU-T standard, seems reasonable for most local multichannel applications, and as in/out format to routers. No final resolution made.

22.4 Further activities

22.4.1 Topics for future work

A number of topics for future discussions were pointed out (not sorted):

- Simple control protocol for amplifiers, machines, converters++.
- Signalling, format setup.
- Test procedures.
- Positioning of data elements inside the cells.
- Error detection & handling.
- Handling of delay differences > 0.5 mS.
- Physical interfaces.
- Routerless operation.
- QoS parameters.

22.4.2 Future work in the group

The participants decided to keep in touch, and probably arrange a new meeting within some months. But all agreed that the standardisation work must be handed over to a more formal body, preferably AES.

22.4.3 AES Activities

The AES-X92 project.

On initiative from Lawo, an AES project X92 Digital Audio in ATM is already established [27]. Little work is done so far. But on the September 1999 meeting of Working Group SC-02-02, a task group SC-02-02-E was set up.

Another related project, X94 Professional Audio via Synchronous Optical Network, is also started. A proposal is presented by Mr. T. Thompson. He will also participate in X92.

AES Task Group SC-02-02 E

Chairman of the Task Group is Mark Yonge (SSL), who also participated in the ATM Symposium. The initial task of the group is *"to produce a requirements specification and liaise with any other bodies working on this topic"*.

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[25] AES Technology Report TC-NAS 98/1: "Networking audio and music using Internet2 and next-generation Internet capabilities". AESWP-1001, Audio Engineering Society 1998.

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[27] AES107 SC-02-02 Report.

23. HDTV Over Internet2 Networks

23.1 Overview

Engineers at the University of Washington, working with colleagues at Stanford University and Sony Electronics, have developed technology to send studio-quality high definition television signals over the Internet. The demonstration project, developed in support of the ResearchTV consortium, differs from previous efforts to send HDTV over data networks in that it does not utilize dedicated circuits or ATM network technology. Rather, this effort relies exclusively on Internet technology and the capacity of the "Abilene" Internet2 research network[1].

23.2 Background

In February 1999 ResearchTV and the University of Washington conducted one of five high-bandwidth experiments selected by Internet2 developers to successfully test the coast to coast connectivity of "Abilene," an Internet2 backbone network. In that case, the goal was to send 7 Mbps MPEG-2 (standard definition broadcast quality) research video streams from the University of Washington in Seattle to Union Station in Washington, D.C., the site of Internet2's inaugural celebration.

This latest demonstration pushes the performance frontiers of the Internet2 research network even further by sending high definition television streams at 40 Mbps and 270 Mbps. This project will give Internet2 developers crucial information about the behavior of the network under heavy load, as well as opening new frontiers for the broadcast industry. This is consistent with the primary objective of Internet2: to enable development of advanced applications that would not be possible without the capabilities of the Internet2 Abilene backbone network.

23.3 Goal

The project goal is to demonstrate the feasibility of sending continuous streams of broadcast-studio-quality high definition (HD) video over a general purpose, multi-user IP network. In particular, from Stanford University to University of Washington, via Internet2¹s Abilene backbone, while maintaining perfect high definition picture quality.

Notably, for this experiment, the Internet2 capacity will not be reserved or pre-allocated, nor will any Quality-of-Service (QoS) or packet prioritization mechanisms be used within the network. That is, the HD video packets will be contending for bandwidth along with other Internet2 traffic. Beyond proving the feasibility of using only Internet technology for HD video transmission, it is also a goal to understand the effect of high-bandwidth applications on other traffic, and vice versa.

The project encompasses two distinct efforts, both based on broadcast industry standards. First is to send a 40 Mbps DVB-ASI stream; second is the more ambitious goal of sending a Sony HDCAM®/SDTI stream at 270 Mbps.

23.4 Process

For both demonstrations, Stanford University is the originating site, and University of Washington is the destination site. Commercially available equipment is used to originate and display the HD video streams; what was missing for this project was a device to take

[1] For more information about Internet2, see section 24.

standard HD streams and turn them into Internet datagrams. To solve this problem, University of Washington engineers combined specialized video hardware and high-performance personal computers with original software developed over the past several months. This code is designed to strike a balance between the number of data packets sent, the size of the data packets, and the type and amount of error correction included. The requirement is to be able to recover from packet loss levels encountered in typical network conditions.

In all cases, a Sony HDW-700 high definition video camera is the source of a digital 1080I HD signal carried over an HD-SDI 1.5 Gbps link. For the 270 Mbps SDTI test, a 994 Mbps source video payload carried over HD-SDI is fed into a Sony HDW-500 VTR. The VTR compresses this video payload approximately to 140 Mbps for the video using Sony¹s proprietary HDCAM compression system. The output of the VTR is HDCAM-rate video embedded in an SDTI 270 Mbps (SDI) stream. This SDTI output is the data format used as the input to the computer which prepares the stream for transmission over the Internet. This computer adds error correction before placing the video frames in Internet Protocol/User Datagram Protocol (IP/UDP) packets and then sends those packets out the Gigabit Ethernet interface card in the workstation.

In the 40 Mbps ATSC compressed stream, the same HD video feed that is used for the 270 Mbps link is applied to a General Instrument HDTV encoder. The 1.5 Gbps HD video is compressed to a 40 Mbps ATSC-compliant serial data stream, carried over a DVB-ASI interface, in MPEG-2 format, which is sent to the network encoding computer. Although different PC hardware and software is required to handle this stream, as in the 270 Mbps case, the PC is responsible for adding error correction and encapsulating the video data into IP/UDP packets which are then output on the workstation¹s Fast or Gigabit Ethernet interface.

When the HDCAM tape is played in the VTR, the video over both links switches from camera to pre-recorded material.

Both of the sending workstations are connected to high-speed Ethernet ports on the Stanford University network, which in turn is attached to the California Research and Education Network (CALREN) and Internet2 via an OC-12 (622 Mbps) Packet-Over-SONET connection. Between Stanford and UW, the data traverses ten nodes, including CalREN-2, Abilene, Pacific Northwest Gigapop, and the UW campus network. The Abilene Internet2 link between northern California and Seattle is a packet-over-SONET OC-12c connection carrying general Internet traffic from the three hundred Internet2 consortium members. The various networks include routers and switches from multiple vendors.

Upon arrival at the University of Washington, packets are received by another pair of specially configured PC workstations. The 40 Mbps stream is recovered and formated into a DVB-ASI transport protocol that goes into an industry-standard GI HD MPEG decoder and a Sony HDM series 1080I high definition display. In the case of the 270 Mbps stream, an external SONY HDCAM decoder is required. The output from the HDCAM decoder is then applied to the high definition display.

No ATM technology was used for this project, and no part of the network was dedicated to or reserved for these tests ‹ the HD video stream was sent over a general purpose, multiple-access network. The general configurations at each site are depicted in Figure 5 and Figure 6.

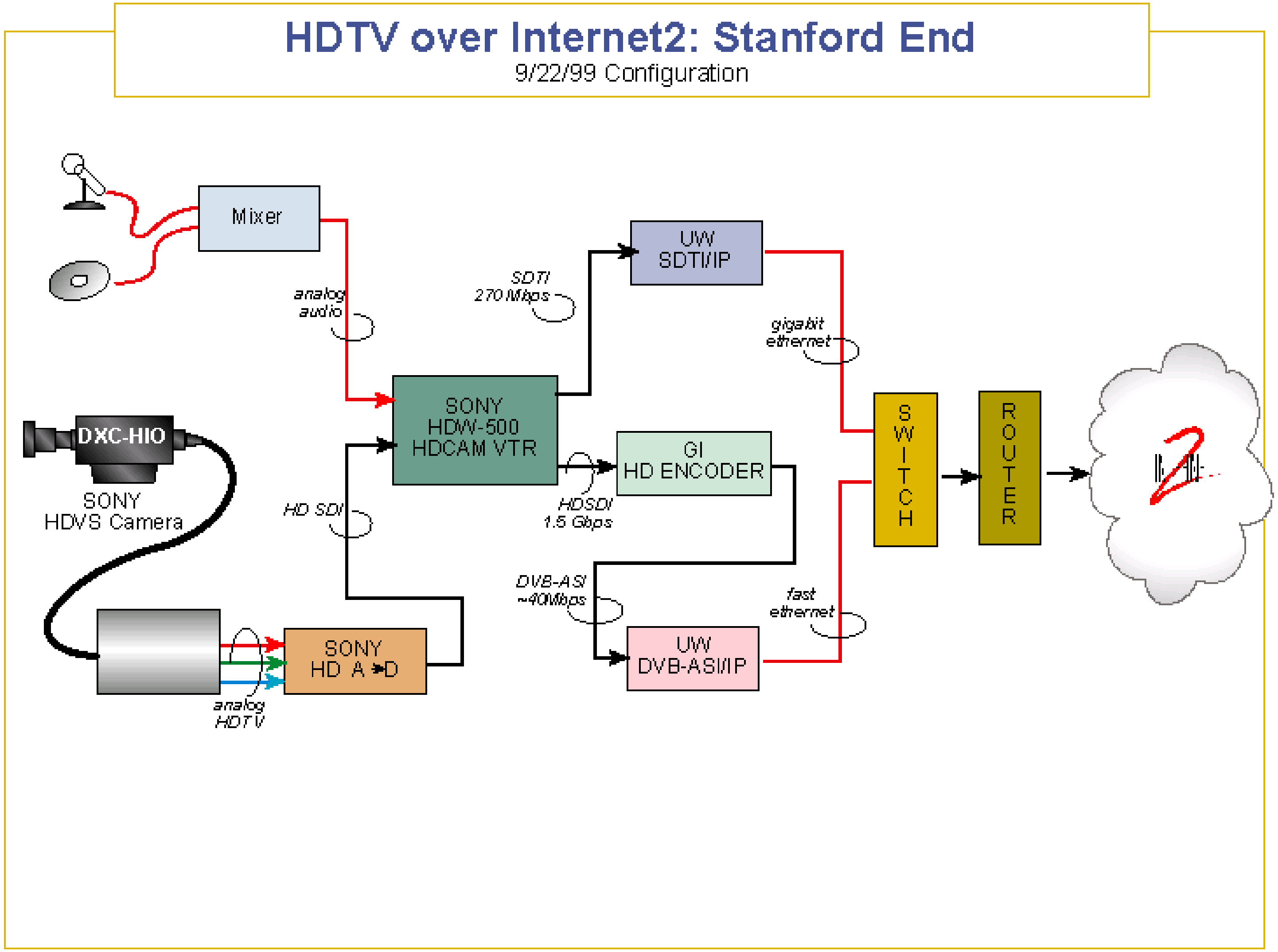

Figure 5: HDTV over Internet2: Stanford End

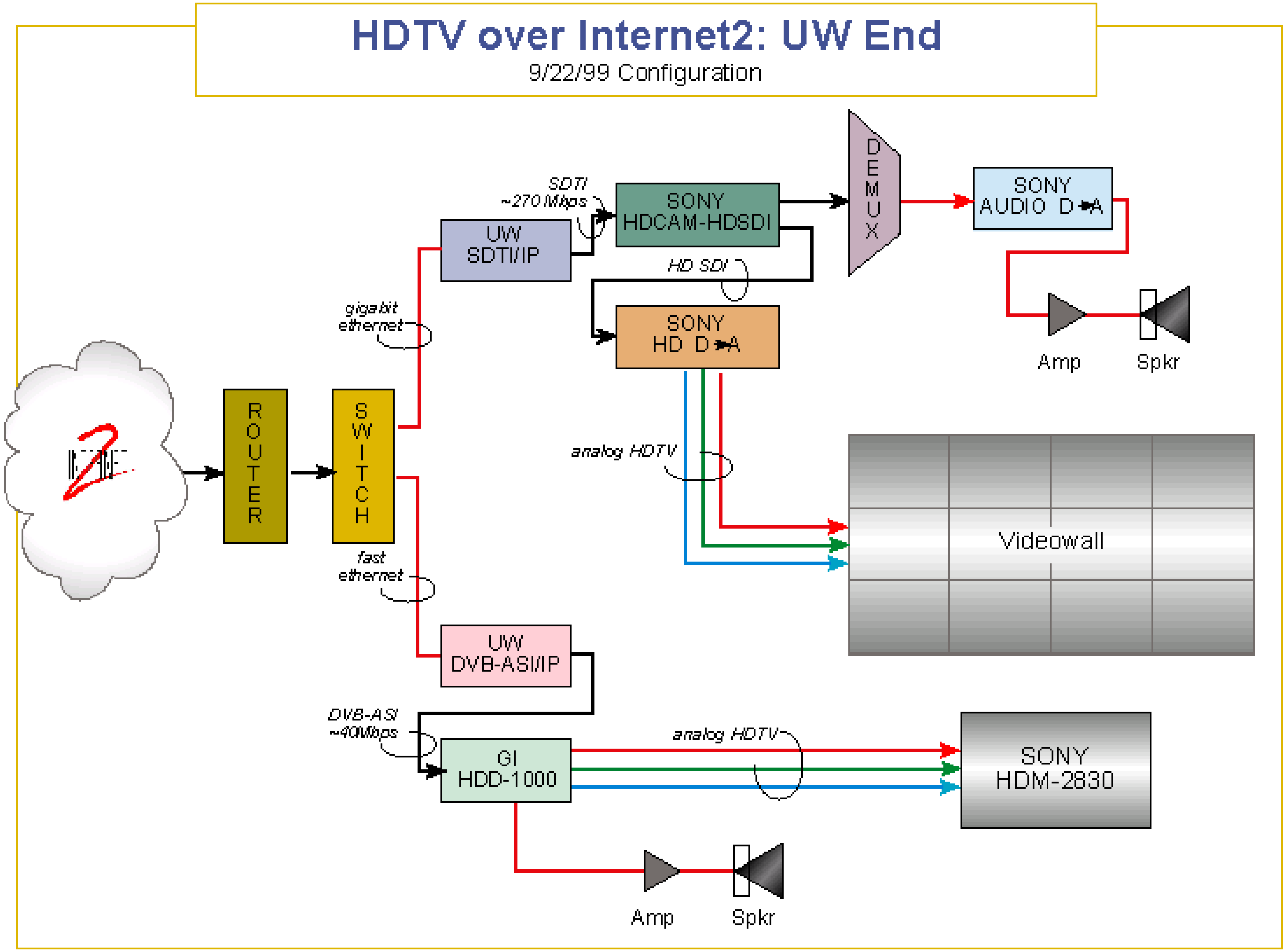

Figure 6: HDTV over Internet2: University of Washington End

23.5 The ResearchTV Consortium

ResearchTV is a collaborative partnership of research universities and corporate research centers dedicated to broadening access to and appreciation of our individual and collective activities, ideas, and opportunities in basic and applied research. The ResearchTV consortium was formed in 1996 to create greater access to high bandwidth research information through technology collaborations with the goal of informing the public about the research that affects the future developments of our world. ResearchTV institutions bring a wide variety of subjects to the public, providing a public forum for researchers to describe their latest endeavors.

The ResearchTV consortium consists of the country¹s leading research universities and institutions: Carnegie Mellon, Duke, Princeton, MIT, National Institutes of Health, Rice, Stanford, University of Alaska-Fairbanks, UCLA, UC San Diego, University of Hawaii, University of Pennsylvania, University of Texas at Austin, University of Virginia, University of Washington, GTE, IBM Corporation, ICOS, and Sony Electronics Inc. University of Washington serves as the consortium leader and administrator. More about Research TV can be found in section 0.

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24. Internet2 — an Overview

24.1 Mission

Facilitate and coordinate the development, deployment, operation and technology transfer of advanced, network-based applications and network services to further U.S. leadership in research and higher education and accelerate the availability of new services and applications on the Internet.

24.2 Goals

- Enable a new generation of applications
- Recreate a leading edge research and education network capability
- Transfer new capabilities to the global production Internet

Some additional specific objectives have been set forth:

- Demonstrate new applications that can dramatically enhance researchers' ability to collaborate and conduct experiments,
- Demonstrate enhanced delivery of education and other services (e.g., health care, environmental monitoring) by taking advantage of "virtual proximity" created by an advanced communications infrastructure,
- Support development and adoption of advanced applications by providing middleware and development tools,
- Facilitate development, deployment, and operation of an affordable communications infrastructure, capable of supporting differentiated Quality of Service (QoS) based on applications requirements of the research and education community,
- Promote experimentation with the next generation of communications technologies,
- Coordinate adoption of agreed working standards and common practices among participating institutions to ensure end-to-end quality of service and interoperability,
- Catalyze partnerships with governmental and private sector organizations,
- Encourage transfer of technology from Internet2 to the rest of the Internet, and
- Study impact of new infrastructure, services and applications on higher education and the Internet community in general.

Begun in October 1996 by 34 US research universities, Internet2 today has over 140 member universities which are working with corporate and affiliate members.

24.3 Member universities

Faculty, staff and students at Internet2 universities are carrying out much, and form the largest part, of the Internet2 project. In joining the Internet2 collaborative effort, the chancellor or president of each university has committed significant resources towards upgrading campus networks, connecting to regional gigaPoP efforts, and the providing the facilities for advanced application development.

24.4 Corporate partners

Corporate partners have committed to supporting the efforts of member universities through close collaboration, including contributions of more than $1 million over the course of the project. Industry participation is key to realizing Internet2's goal of broad diffusion of advanced networking capabilities. Corporate sponsors and corporate members are also working with the university and affiliate members of the Internet2 project.

24.5 Affiliate members

Affiliate members are organizations with a strong interest in the mission and goals of the Internet2 project. Many affiliate members are providing leadership in developing the Internet2 gigaPoPs.

Internet2 Working Groups have been established to explore specific technical challenges related to enabling advanced network applications. Some of these challenges include implementing scalable quality of service, IPv6, and multicasting. Working group members include representatives from Internet2 universities, as well as affiliate and corporate members.

Internet2 Initiatives such as I2-DSI (Internet2 Distributed Storage Infrastructure), I2-DVN (Internet2 Digital Video Network) and the QBone (Quality of service Backbone test bed) have been undertaken to explore and deploy new network technologies needed realize the Internet2 project's goal of enabling advanced applications.

A compact central staff supports the efforts of individuals from the university, corporate and affiliate members. With the rapid growth of the Internet2 project, the University Corporation for Advanced Internet Development (UCAID) was established in September 1997 to provide an organizational home for the effort. UCAID, along with its partners, undertook the Abilene Project in support of Internet2 in April 1998. UCAID's Board of Trustees and Advisory Councils provide guidance for the Internet2 project.

Some links:

http://www.internet2.edu/html/news.html

http://apps.internet2.edu/talks/

25. ResearchTV — an Overview

ResearchTV is a collaboration founded by a core group of leading accredited research universities and research organizations to experiment with opportunities to expand high bandwidth modes of delivery and exchanges in educational and research-oriented information.

Originating in 1997, ResearchTV participants began plans to develop projects with the goal of facilitating new methods of national and international communication about research information for use interinstitutionally and for the public.

25.1 Introduction

The following is the original Research TV proposal as written by the University of Washington and IBM, January 1997. This proposal served as an introduction of the Research TV concept and helped to initiate discussions with research institutions and organizations.

We propose to form a collaborative partnership of research universities and corporate research divisions dedicated to broadening the access to and appreciation of our individual and collective activities, ideas, and opportunities in basic and applied research.

The basic structure would be a video programming service, developed and operated by this partnership. This video service would effectively communicate the contributions made by research to our educational, cultural, and economic prosperity; and the trends and developments that have influenced or will impact our world. It would feature new ideas, techniques, discoveries, current theories, profiles of research leaders and projects, and vital research applications.

The service would be available to the broadest possible audience: the general public, educational institutions, and businesses via both broadband and on-demand communications technologies in both the analog and digital domains.

25.2 Research and Public Outreach

The complex role of university-based research is easily misunderstood, devalued, and portrayed by the media (and politicians) as more of a boondoggle than a boon. While new consumer products, medical treatments, or auto safety features-applications of research-are covered by the press, the research endeavors behind these applications are rarely front-page news. Instead, university and corporate research is subtly woven through everyday stories, and the participation of universities and corporations in the ongoing economic development of our country is not clearly conveyed to the public and is often taken for granted.

Corporations have been somewhat successful in leveraging their research efforts to market brand-name recognition and products to the general public. Still, many corporate research departments remain hidden deep within these organizations. Invisible as well is the tremendous amount of collaborative research that takes place among peers from educational institutions, corporations, governmental agencies, and research organizations. These collaborations transform the best each organization has to offer-academic creativity, applicability to business and social welfare-into discoveries for the advancement of all of society.

The research community’s efforts to inform the public about research and its role in such areas as education, business opportunity, and technology transfer are limited by funding and ranked as low priorities. In turn, outreach is often limited to articles, op-ed pieces, testimony, and public talks that reach small local audiences.

Research TV would provide an opportunity to broadcast video materials, including symposia, panels, and lectures originally created for the local communities to a much broader national and international audience. Through Research TV, a researcher in San Francisco could provide a rare glimpse of her work to a viewer in the Bay Area as well as to a colleague, student, sponsor, or lifelong learner in Indiana or India. Research TV would improve and create new connections among the public and the research community having a substantial and direct impact both in the public's use of research information and in the public's support for continuing research efforts.

25.3 New Communications Technologies

New technologies are providing research institutions with communication tools to reach the larger community. The Internet, a technology pioneered and developed by universities themselves, and other technologies such as direct broadcast satellite (DBS), are creating new opportunities to develop a voice for research. These new distribution technologies paired with the recent changes in telecommunications law have made this a uniquely opportune time to develop this new voice for research and the institutions that support and conduct research. An infrastructure for distributing video-based materials is emerging with the advent of widely available, affordable broadcast and on-demand communications to the general public and organizations of all kinds.

Universities and corporations are investing in both traditional and developing distribution technologies as the demand for greater access to educational opportunities continues to grow. The concept of "virtual university," now gaining public interest, exemplifies the increasing demand for educational and informational programming. The foundation for a virtual university presupposes a robust, high-speed network that could deliver many types of media on-demand to students of diverse needs, ages, and locations. The coupling of the virtual university with Research TV begins to define a comprehensive educational system for the continuing learner, as well as to provide all interested parties with easy access to the latest research in a field of study.

Public and private organizations see the value of expanding and opening up more distribution channels through such efforts as Public, Education, and Government ("PEG") access television channels; programming via satellite systems; and the movement to on-demand and multicast multimedia distribution via high-speed Internet services (e.g., services over cable systems such as "@home" or newer ADSL technologies available through telephone companies). As these enabling technologies continue to emerge, Research TV will be active in coordinating a shared body of materials from leading research institutions. Through Research TV these materials will be made available to a broad reach of viewing publics providing a steady stream of stimulating, engaging, and informative programming.

25.4 Proving the Concept - The UWTV Model

The idea for "Research TV" and the partnership among universities and corporate research entities resulted from our exposure to universities eager to increase their methods and range of public outreach, and from positive public response to the experiments we developed through the University of Washington television channel, UWTV.

UWTV has developed a high-profile line-up of programs that feature the lectures, demonstrations, and applications of important cutting-edge research. These programs are currently distributed to our local community via television and the World Wide Web. UWTV staff produce video programming at the University of Washington that exhibits the research of various University of Washington faculty and departments. UWTV also collaborates with other universities to obtain video programming that promotes the research of each of these universities to UWTV audiences. Many of our televised programs are supplemented by World Wide Web pages on the Internet that relate associated problem sets, reading materials, and references on the Web. This research-focused programming presents an extremely rich source of infor-mation to the public, and UWTV has come to be known by viewers as a source uniquely dedicated to relaying information about research.

UWTV's collaborations with universities such as Stanford, Harvard, University of Wisconsin, UC San Francisco, UC San Diego, Duke University, University of Virginia, Rice, Massachusetts Institute of Technology, and others have convinced us that universities are expanding their vision for public service, outreach, and public relations and are looking for more powerful ways to bring information about their research efforts and results to their communities.

Not only has UWTV served as a test-bed for such collaborations between universities, UWTV also provides a model to test viewer response. Reaching over half-a-million households across Washington State, UWTV receives a high response rate and positive comments from viewers who watch our mix of lectures and obtain related learning materials. Every week we receive numerous phone calls and email contacts from people wanting to receive TV schedules, to purchase programs, and/or to comment on the high quality of information they have viewed. Viewers report regularly that they are excited to find a station that features original, reliable, state-of-the-art information presented by world-renowned experts.

While broadcast versions of UWTV programs are only seen in areas where they are carried, the listing of programs and materials on the Internet reach a global audience and Internet users around the world often make inquiries about the programming.

25.5 Goals for the Future

25.5.1 Create a Media Presence

To create a regular and accessible voice for research universities in the national and international media.

25.5.2 Public Information and Education

- To open up the "ivory tower" to the public, engage the public in the conversations, developments, realities, and products of university and corporate research missions.
- To reach out to people in their homes and businesses and engage them with ongoing research activities that may be of professional or personal interest to them.
- To demonstrate to the public that research is an integral part of education at all levels, from formal education through professional life and beyond; that research serves as an educational process for students and professors; and that the dynamic connection between research, education, and business is a key value provided to society by a research university.
- To encourage credibility in our own local marketplaces by providing current information about research developments and by demonstrating that the growth of businesses and organizations depends on the dedicated and long-term support of research.
- To regularly educate the public about the importance of the "art" of research, its fusion of creativity, curiosity, expertise, and vision; an ongoing process rooted in our

history of thought, and extending forward into our goals for tomorrow

25.5.3 Encourage Collaborations and Experiments among ResearchTV Participants

- To combine the research efforts of universities and corporate research divisions with new communications technologies in order to build a stronger bridge between the two for the exchange of ideas and the development of new collaborations.
- To experiment with distribution technologies and combinations of technologies in order to enhance the learning process and stimulate the exchange of ideas, collaborations, and technology transfer between education, business, institutions, and organizations globally.
- To use content, content creation, and manipulation processes as a workbench to test materials for our future analog and digital broadcast and on-demand multimedia offerings, thus providing an unusual opportunity to experiment with new methods of distribution and interaction on a global basis.

25.6 Next Step

Representatives of selected universities and corporate research divisions met for the initial Research TV meeting in March of 1997 and agreed to move forward to achieve the proposed goals. Meetings are now being scheduled on a regular basis to explore the creation and development of Research TV.

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