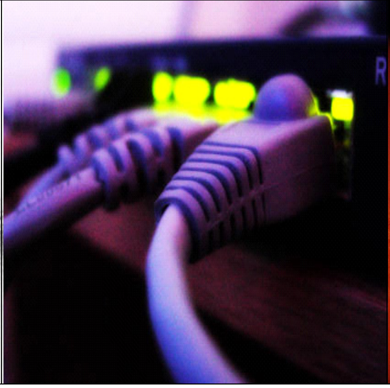
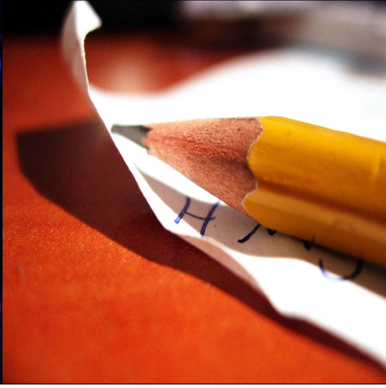

Multiplay Architectures

Building the Convergent Network for Data , VoIP, and IPTV applications

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Preface

The telecommunications industry is undergoing important changes as service providers and users have already moved away from the model of the past century where voice, between fixed telephones, was the essential part of the business. Technological shifts, social changes and competition have persuaded operators and manufacturers to redefine the business model around new multiservice networks capable of delivering voice, TV, and Internet access by means of unified infrastructures. Indeed, the whole world is changing faster than ever thanks to the new transport and telecom technologies that can move people, goods, and information faster than the old galleons did. Harbours and routers are nothing but open windows to the interchange between human communities. The result has always been the same, globalisation, which can be simplified as an increment of trading that is followed by certain degree of cross cultures. Globalisation is a controversial process, which is often perceived as a combination of feelings that swing from the hope and interest, to the fear and concern. Because it is truth that new technologies are also backdoors that may eventually simplify our cultural diversity. Independly of the opinion everyone has, it will not be the first time globalisation occurs.

The history shows outstanding samples when a communications network facilitates the establishment of permanent links first between towns and then between cultures that interact enriching each other. For instance our phonetic alphabet was initially designed by the Phoenicians, then it was improved in Greece adding the vowels that allowed the use of Greek as the first international language; finally these scripts were adopted and reshaped by Rome that spread its use across Europe and the Mediterranean countries. Since then only the lowercases have been invented before writing this book. The Mediterranean sea is indeed a good sample of how a good network generates synergies. Phoenicians, Greeks, Romans, Byzantines, Arabs, Iberians, Ottomans, and French emerged as great civilizations at the rims of this sea, to extend the influence of their culture far away the borders of their home land. From Algeciras to Istanbul, from Nice to Alexandria traders, soldiers, monks and families were sent to the other side of the coast to interchange goods, blood and culture, many times, under the sword pressure. The maritime routes facilitated such a level of interchanges that many cultural and racial borders were blurred, but it is interesting to realize that never produced an homogeneous nation because despite of the continuous changes and transformation, people's identity was kept because there were also forces that generated diversity.

Among all the aspects of the culture cooking is one of the most interesting since are less fastened to the impositions of the secular and religious authorities. Cooking is all about art and techniques, with the epicurean intention of feeding friends and relatives while maximizing the pleasure of eating and drinking. Cooking has been particularly important in the Mediterranean where the kitchen occupies a central place at home. Time slows down for lunch and diner, and invitations to have food at home is the most common expression of hospitality. Cooking recipes are fascinating. No cuisine is established in isolation, what means that cooking in Egypt is not a carbon copy of that found in Lebanon, or in Greece although all clearly show the Mediterranean influence. In a recipe you can trace not only climate and taste preferences, but also trading routes, migrations, invasions and many cultural aspects related with the calendar and believes. Mediterranean cooking styles change across regions. You may find many variations of the same recipe, even seamless transitions between two recipes apparently different. In any case there is always something that allow you to identify them as Mediterranean which is the result of a crowded history of lends, and borrows that the maritime network made possible.

The first civilizations, Phoenicians, Greeks and Romans, established the trinity of products, that is, olive trees, vineyards and wheat. For generations olive oil, wine and white bread have been signatures of the Mediterranean cooking, totally embedded onto their lives and traditions, and leaving permanent footprints in all the civilizations developed on its shores.

Just two centuries after the Roman Empire collapsed Arabs begun the expansion over the North of Africa and the South of Europe. They were great traders that introduced rice, sugar cane, oranges, grenades, spinachs and aubergines that came from remote places like India and China. Extraordinary samples of the Arabic heritage is Couscous which in Sicily is generally combined with fish, and, candies like marzipan, panellets or turroni, very popular in places like Alacant, Barcelona, Marseille or Cremona which are nothing else but local versions of the outstanding variety almond based sweets that you can still enjoy in the medinas of Fez or Marraquech.

The Spanish expansion in America added new ingredient to the cooks including beans, peppers, potatoes, maize, and tomatoes. Those territories integrated in the Aragonese crown quickly adopted the new vegetables that could grow easily on their fields. Many recipes were transformed like the paella in Valencia that would not exits without the green beans; or the modest polenta in Italy, which was definitely improved by substituting the barley flour by corn flour. Would anyone imagine today Venice without grilled Polenta with calamari al nero?

We can't forget the Ottomans that dominated most of the East and North Africa for several hundreds of years since Byzantines were defeated in the XVth century. This also was an interesting case of melting pot, since this empire granted all type of exchanges. You should be an expert to distinguish the differences of the small puff pastry cakes that you can have either in Essahuira, further west in Morocco, either in Athens in front of the orthodox Cathedral, either in the narrow streets of the old town of Jerusalem.

There are many more, but Sephardim are protagonists of one of the most fascinating stories occurred in the Mediterranean. It is believed that Sephardim Jews arrived and lived in relative tolerance in Iberia since the era of King Solomon (930 B.C.E.). But in 1492 the expulsion edict was signed in Spain, for those who rejected to be converted to Christians. Large Sephardic communities were founded in Amsterdam, Geneva, London, Bordeaux, and Hamburg. Nevertheless the majority preferred to look for a new home in the Ottoman territories whose sultan Beyazit welcomed them. For centuries Sephardims preserved their culture, including the jews religion, their language, which is a combination of Hebrew and Spanish and, of course, their cooking style which is a combination of the Iberic heritage and the Turkish, Greek or Arab flavours that were introduced in a gradual osmosis. The Sephardic kitchen is very Mediterranean and relies on appealing combinations of meats, vegetables, or fish served up in casseroles, pies, stuffed vegetables, or pastries. Baking foods, deep-frying, or simmer on top of the stove. Still today you may have mostachudos or huevos haminados, in the juderias of Tanger, Sofia, Thessalonica, Istanbul, and Sarajevo. The deep-fried fish technique was introduced by them in Amsterdam after the expulsion, and then to England where initially was known as the jews way of cooking fish. Finally, in 1860, Joseph Malin opened the first fish-and-chip shop in London. Except the olive oil and the squeezed lemon, which was substituted by malt vinegar, the recipe is today exactly as it was five hundred years ago. Curiously deep-fried was also adopted in Japan when Portuguese traders and the jesuit Francisco Xavier mission settled in Nagasaki in the XVIth century. Tempura, the popular Japanese dish, is a well documented legacy of that time, in which seafood and many different types of vegetables are coated with batter and deep-fried in high *temp*era*tura* oil. Surprisingly two of the most peculiar British and Japanese dishes have a common root anchored in the Mediterranean rims.

We have used the Mediterranean to highlight how a network, does a matter if based on ships or in IP packets, facilitates all kind of exchanges between peoples. Multiculturalism is indeed is part intrinsic of the human being. Nothing last forever, we are nothing else than just a picture of a continuous transformation, but this fact does not necessarily means uniformity in the globe. We want to be optimistic, we want to say that communication, in the widest sense of this word, means cultural enrichment and more opportunities to understand, and to enjoy the differences of the world. It should be just a matter combining properly the new ingredients with those that are part of our backgrounds to maintain our peculiarities.

Regarding this book (which is about communications, not about cooking), just to remark how innovations in telecom have produced such a significant changes in our lives in a very short period of time. Now we talk about personal and mobile communications, and broadband Internet access which is more flexible than ever, thanks to the technological convergence that have permitted the bundle of television with the existing data, and voice services. The convergence of IP centric applications over unified infrastructures can generate significant benefits, but it is important to assure that performance and quality parameters. Testing these networks, the availability of the services, and early detection of any issue that may affect the customers' experience. The audiovisual market is very mature, it has been served by broadcasters for years, therefore it is of key importance to match customers expectations from the first day pay. Experience demonstrates that trial and error strategy does not work in consumers market.

The authors

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Chapter 1

Business Strategies

Telecoms operators are now employing new strategies to deliver thrilling new services using next generation networks. The full package of services includes line rental and fixed line telephony with a combination of Internet access, IP television, video on demand, entertainment applications and, eventually, cellular phones. Using the terms adopted by the industry, we are talking about *Triple Play* and *network convergence* in other words this means: multiple services, multiple devices, but one network, one vendor and one bill (see Figure 1.1).

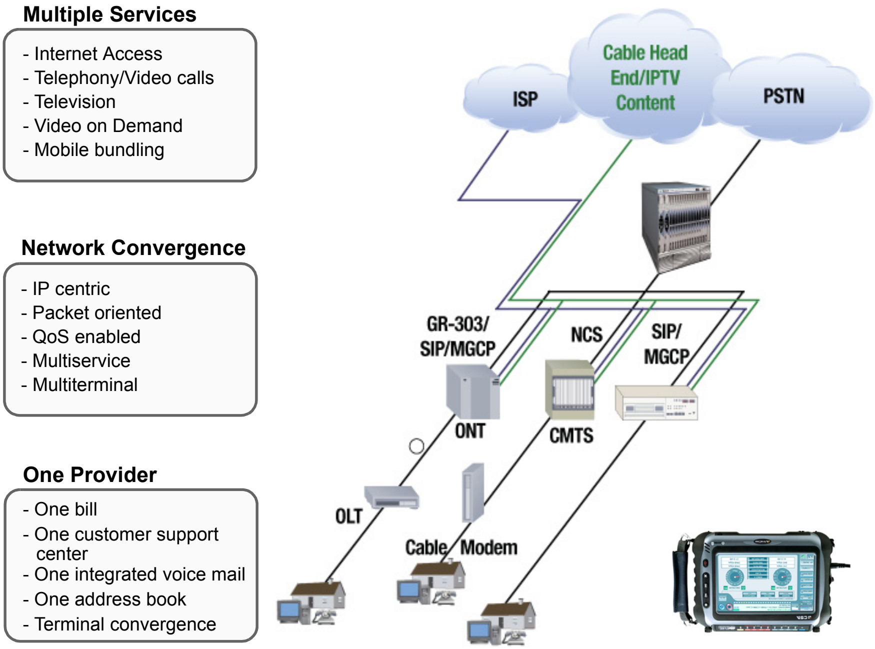
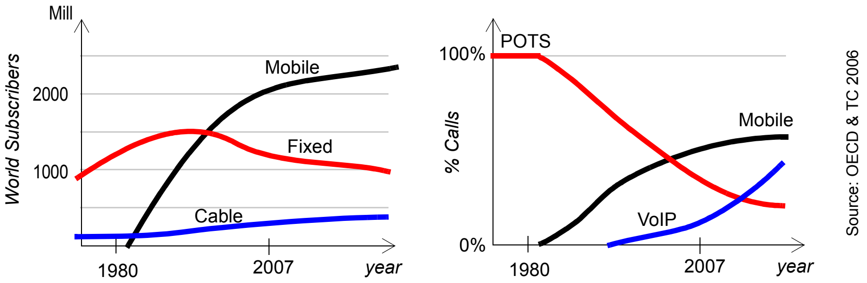


Figure 1.1 Triple Play aims to unify telecommunication services by using a single network to deliver a package of services.

This maneuver is much more than just a new commercial product. It is a consequence of the important changes the industry is undergoing such as technological innovations, social changes and new regulations. These changes have persuaded operators to redefine their businesses based around a new unified network that should be able to support any type of telecoms service.

Beyond each particular strategy we can identify some of the common drivers such as a declining voice business (see Figure 1.2), the flat profit perspectives on data access, the new regulations encouraging competition, and the technological achievements that have made network convergence possible.

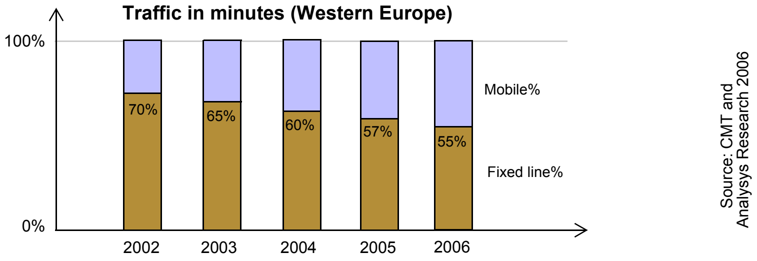


Source: OECD & TC 2006

Figure 1.2 Mobile and cable subscribers undermine up to 2% of fixed line subscribers per year

1.1 EXPANDING TELCO BUSINESSES

Voice is still a profitable business with margins of over 50%, however, in the case of traditional fixed telephony, it is rapidly declining (see Figure 1.3). In many cases the fixed phone service is included in the same flat bill as the DSL access and line rental. On the other hand, the growth rate of the cellular telephony business is now less than it was a few years ago and is nearly at saturation point in the developed countries. Unfortunately for fixed line operators, alternative services such as broadband access, have quickly become difficult to define, making it impossible to compensate for declining voice revenues.



Source: CMT and Analysys Research 2006

Figure 1.3 Voice revenues based on fixed lines is declining and wireless voice revenues are not growing as in the previous decades.

After several failed attempts the telecoms industry has, apparently, found a remedy for its continuous headache of offering multiple services as a commercial package that includes fixed line rental and wireless services all in one monthly bill (see Figure 1.4). These multiple services are often referred to as Triple Play. It is too early to know if this is going to be the solution for the 3G crisis that started in 2000 or is it just 'another case of mass hysteria in the telecoms industry'[1]?

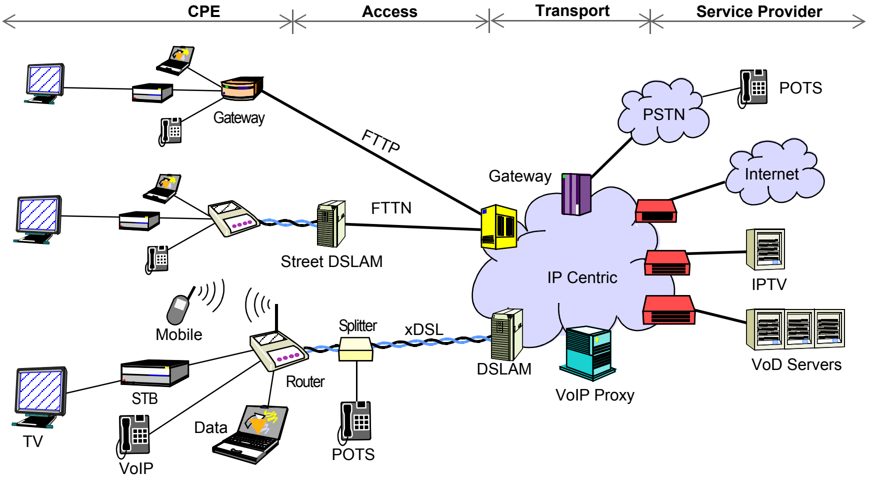


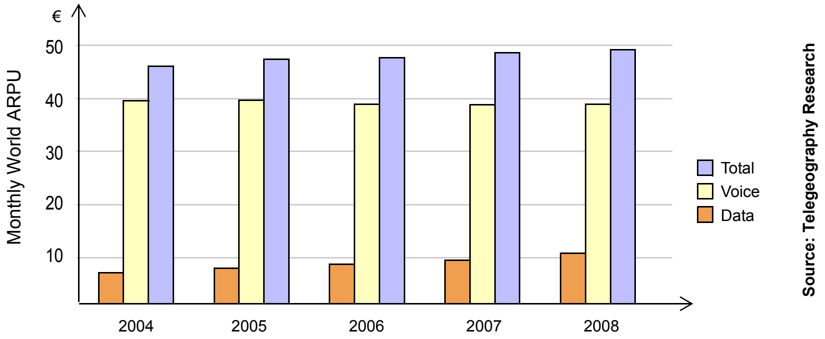
Figure 1.4 Multiple services means one network, various terminals and many types of access.

Bundling video with existing data and voice services goes further than a pure marketing campaign; it is essential to keeping telcos in the residential business. It is, in fact, a very ambitious strategy with well defined targets:

- *Reduce Churn*. Gain customer loyalty with one package that includes all services supplied by one vendor.
- *Minimize Costs*. Integrate infrastructures and human teams using network convergence.
- *Gain TV customers*. Telecom operators should use the same weapons as the cable companies to supply television services.
- *Increase Profits*. By using legacy and innovative applications to raise the average revenue per user (ARPU) (see Figure 1.5).
- *Grow the brand name*. Cultivate the perception of the company as being able to supply any type of telecommunication service.

[1] The Economist magazine. 12th of Oct, 2006.

- *Improve Service provision.* Use advanced management solutions for quick and easy provisioning.
- *Network Convergence.* Incorporate new technologies and recycle existing network infrastructures.



Source: Telegeography Research

Figure 1.5 Evolution of world ARPU.

Triple Play is focused on a combined service rather than on the development of new applications, protocols or architectures (see Figure 1.8). We have already explained this is as a marketing concept concerning: Internet access, television, and phone services. All of them provided by one vendor, delivered over a single access network and paid with one bill. To offer Triple Play requires a technologically enabled network that can transport all the three basic communication flows (audio, video, data) through the same pipe. In essence Triple Play is not really a new service, but a concept related to a new business strategy on two planes:

1. **Commercial bundle,** the concept referred to as a package of services.
2. **Technological convergence,** the concept referred to as an IP centric network, enriched with the facilities to support and deliver all the services.

Triple Play is appealing to customers because it simplifies the subscription and support of several telecoms services. The problem for service providers is that price, quality and contents are very important, especially as most services are not new at all, except in the format and the interfaces that are used to supply them. Aims to reduce customer bills would also reduce total market revenues therefore development of new services is essential (see Figure 1.6).

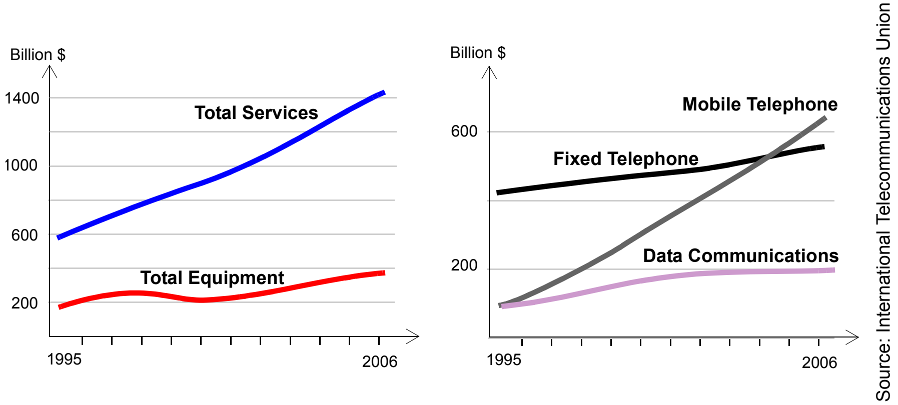


Figure 1.6 Telecom World market revenues in Services and Equipment.

1.2 Triple Play Applications

A large number of applications can be designed and supplied over a converged network (see Figure 1.7). Triple Play applications are often a combination of several types of information, and a number of parameters such as bandwidth, source/destination relationship, type of routing, QoS, and traffic symmetry (see Table 1.1).

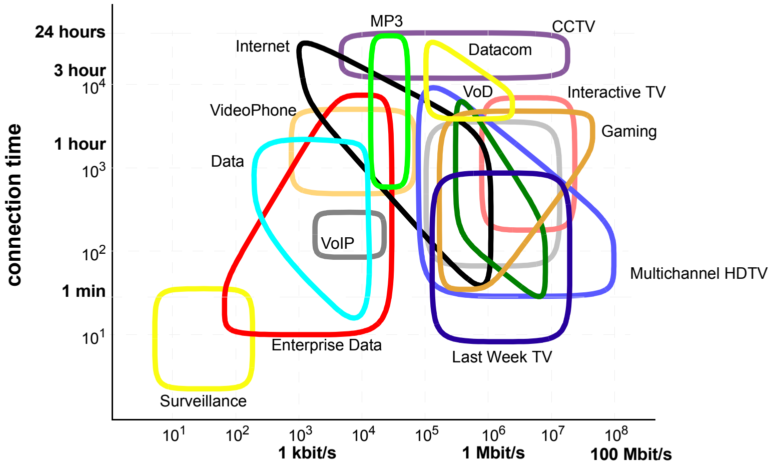


Figure 1.7 Triple Play applications: Bandwidth and time requirements.

Table 1.1
Triple Play applications.

Application	*Pld*[i]	*Rel*[ii]	*QoS*[iii]	*Sym*[iv]	*Comments*
IPTV Channels	A, V, D	P2M	P	A	Free of charge
HDTV	A, V, D	P2M	P	A	High Definition IPTV
Pay per View TV	A, V, D	P2M	P	A	Pay per view
VoD on STB	D	P2P	B	A	Downloaded to the STB before visualization
VoD on Network	A, V, D	P2P	P	A	Requires a Network Server
Channel Search	D	P2P	B	A	On: theme, actor, language...
Pseudo VoD	A, V, D	P2M	P	A	N delayed channels/program
Video conference	A, V, D	M2M	G	B	Multiparty conference with image
Voice over IP	A, D	P2P	G	S	Inc. data services that is, presence
Broadcast Radio	A, D	P2M	P	A	Using TV transport stream
Streaming Radio	A	P2M	P	A	Using Internet
Voice Mail	D		P	A	Non-real-time voice messages
Hi Fi audio	A	P2P	P	A	Pay per listening session
Audio downloading	A	P2P	B	A	For MP3 players
Gaming individual	D	P2P	G	A	Individual or Group
Gaming group	D	M2M	G	S	Pay per play
Fax	D	P2P	B	A	Fixed to VoIP
e-commerce	D	M2P	B	A	Web sales applications
VPN services	D	M2M	P	S	Home working
Hi-Speed Internet	D	P2P	B	A	Entertainment, Home working
Storage services	D	P2P	G	A	Home working
Surveillance	D	P2P	B	A	Alarms
Home automation	D	P2M	B	A	Remote control, monitoring
Instant Messaging (IM)	D	P2M	B	B	Real-time short messages
e-mail	D	P2M	B	U	Non-real-time messaging
www	D	P2P	B	A	Information browsing based on hypertext
File Transfer	D	P2P	B	A	Data download and upload from/to a server
UMA	A, V, D	P2P	G	S	Unlicensed Mobile Access
Mobile Convergence	A	P2P	G	S	Call redirection to fixed line
SMS	D	P2P	B	U	Non-real-time wireless short messages

i. Payload: Audio, Video, Data (A/V/D)
ii. Relation: Point to Point, Point to Multipoint, Multipoint to Point, Multipoint to Multipoint (P2P / P2M / M2P / M2M)
iii. QoS: Best effort, Prioritize, Guarantee (B/P/G)
iv. Traffic symmetry: Unidirectional, Bidirectional, Asymmetric (U/B/A)

1.2.1 Television and Video Services

Television services can be implemented following several models by taking into consideration parameters such as resolution, coding, and the service model. Nevertheless, it is the transmission mode, broadcast or multicast that modifies the service:

- **Broadcast channels**. Channels are broadcast/multiplexed simultaneously in TDM or FDM over the transmission media. Subscribers use the tuner on the

TV box to select which one to display. This model is used by cable, terrestrial and satellite broadcasters.

- **Unicast/Multicast channels**. Channels are streamed independently to reach the customer premises that have selected the stream previously. This is the model selected by telcos, in principal, because of the lack of bandwidth at the first mile.

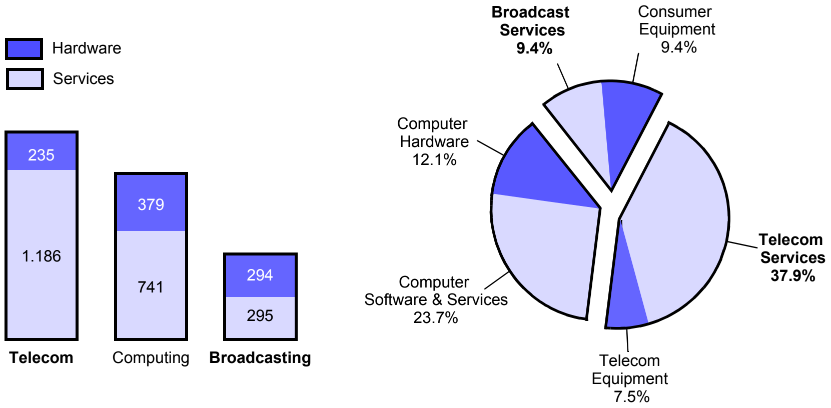
Digital Video provides a set of interesting possibilities such as metalanguage programs (one video, several audio signals), customized adverts, pay-per-view, or encrypted programs only for subscribers. Interaction between the subscribers and the service provider make new capabilities such as games, magazines, voting, competitions, pay-per-view, customized adverts, and quizzes possible. At the end of the day integration is the key difference between unidirectional and bidirectional digital platforms.

Figure 1.8 Residential customers are focused on applications such as IPTV, VoD, video recording, telephony, Internet access, gaming, hi-fi Audio, home automation, and mobile bundling. While business customers, are focused on connectivity applications such as VPN, Broadband access, corporate VoIP, and mobile convergence.

1.2.1.1 Welcome to the Contents

For most telcos, the television and video business is new. Any previous experience of these services was no more than signals transported in SDH envelopes between the different centers of TV broadcasters. But now that Telcos are also service providers, it is necessary to manage, not only the transport and signal distribution, but also the contents; a set of attractive programs to compete with existing cable operators and broadcasters.

Therefore, telcos must acquire, not only new technical and business skills to enter into this already mature market, but should also be involved in contents creation that is adapted to their specific consumer market. This explains why many telcos have created their own studios or signed contracts and joint ventures with content providers to gain access to suitable and appealing programs, libraries of movies and specialized channels.



Source: International Telecommunications Union

Figure 1.9 The global Information and Communication Technologies Market (ICT) indicates that the total broadcasting market is about the 40% of the telecom market, however if we consider only the services segment the size is only the 25%.

1.2.2 Video on Demand

The Video on Demand (VoD) service is quite different to IPTV as it enables users to select and watch a video as part of an interactive system. VoD systems have two important features:

1. They enable users to choose the video they want to watch from a digital library selection. Users can control the moment they want to start watching the video.
2. They provide typical DVD functionality such as pause, fast forward and fast rewind.

For streaming systems this requires more bandwidth on the part of the server, powerful multicast nodes, spare bandwidth and guaranteed QoS control. VoD servers can operate in two ways: streaming or downloading the contents, in both cases this is point-to-point. If downloading is being used only ‘best effort’ is necessary since the video is recorded onto a network disk, a PC, or a set top box before it is watched.

VoD is one of the killer applications that makes the Triple Play service more attractive when operating over a rich IP network, since Cable and Satellite operators have more difficulties in implementing this.

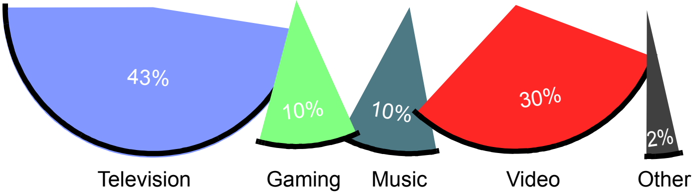


Figure 1.10 Watermelon distribution of the entertainment business.

1.2.3 New TV Receivers

The migration from analog and standard definition TV to digital and high definition TV (HDTV) started several years ago. Now mature markets already have many TV receivers enabled to receive digital signals. The new trends are incorporating High Definition into new receivers and enabling interactivity in commercial TV applications.

1.2.3.1 Coding

The digital codification of a high resolution TV program generates a 20 Mbit/s stream while standard resolution generates about 6 Mbit/s. This can be reduced significantly using compression algorithms such as Moving Pictures Experts Group (MPEG) or Windows Media (WM). Both can offer several alternatives depending on the resolution, and the compression level (see Table 1.2).

Table 1.2
Standard and High Definition bandwidth after compression

	Lines	*Pixels*	*Broadcast*	*MPEG-2*	*MPEG-4*	*WM9*
SDTV	480	704 x 480	6 Mbit/s	3.5 Mbit/s	2-3.2 Mbit/s	2-3.2 Mbit/s
HDTV	1080	1920 x 1080	19.2 Mbit/s	15 Mbit/s	7.5-13 Mbit/s	7.5-13 Mbit/s

The most popular compression family of standards is probably MPEG, defining algorithms based on the ‘cosinus transform function’ that discards spatial redundant information, and employs movement compensation techniques to minimize temporal redundancy. MPEG-2 is very common, MPEG-4 is a step ahead standard that covers small mobile hand sets and up to large HDTV receivers. Windows Media 9, a Microsoft development, is also an interesting alternative.

1.2.4 Voice over IP

In many aspects VoIP is also an approach to the Internet world using IP packets to carry the voice signals to incorporate features that would be difficult using traditional phone services. For example, VoIP allows users to talk for as long they like, they can always be connected and users will always know that they are connected. Users can send images, data and videos simultaneously to the people they are talking to. Another interesting aspect is how VoIP phones can use the e-mail address as an identifier and can make calls to an e-mail address as well. Moreover the phone call list can be made up using a combination of PSTN numbers and e-mail addresses. Access of nomadic users, is guaranteed in a similar way to Internet based mailers, regardless of where in the world the connection to the Internet is established, thanks to proxy servers.

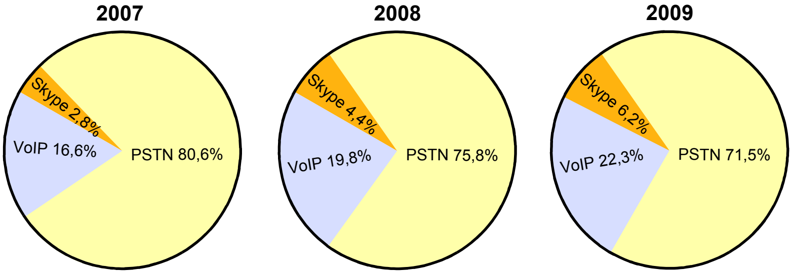


Figure 1.11 International traffic transport.

A commercial VoIP service should not be restricted to VoIP phones, but heterogeneous calls between VoIP and ISDN, POTS or GSM phones will be quite normal for a long time. It is not realistic to forecast a full substitution in the short or middle term.

1.2.4.1 Less Expensive Phone Service

The phone service based on VoIP generally costs less than the equivalent based on the traditional Public Switched Telephone Network (PSTN). This fact has been justified by saying that packet oriented technologies are more efficient that circuit oriented ones because they permit the utilization of a single network to carry voice together with data and video. This hypothesis has been proved, at least according to the 2006 survey carried out by Consumer Reports in the US. The survey said that people who have purchased VoIP service are reportedly saving around $50 per month on their bills.

It is also important to bear in mind that incumbent operators already have a satisfactory quality phone service and in many cases it has already been bundled with ADSL and line rental. That means that the PSTN cost has been deflated and this can explain why legacy operators have little interest in rolling out a new VoIP service that would demand more resources for very low benefits (see Figure 1.12).

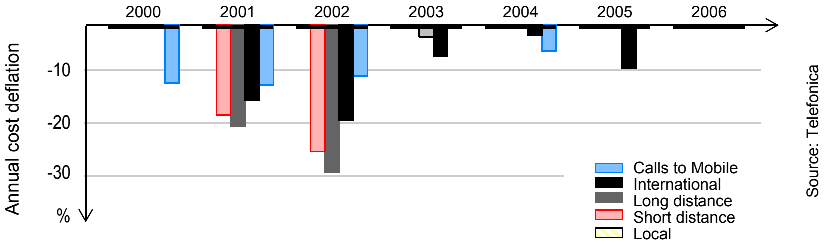


Figure 1.12 Annual deflation of phone calls originated on fixed lines. The Telefonica case.

1.2.4.2 Drawbacks

The VoIP phone service relies on the Internet connection so consequently the service can be affected if the quality and bit rate are not appropriate, causing distortions, noise, echoes and unacceptable delays. Most of the drawbacks can be overcome by increasing the bandwidth, minimizing contention, and using more elaborate protocols to prioritize VoIP traffic while controlling delay and packet loss. It may also be worth considering modifying firewalls and adapting Network Address Translation (NAT) tables used at the boundary routers.

1.2.5 VoIP Rollout

VoIP is a very important technological, financial and social change after a century dominated by national operators and a phone service based on circuits (see Figure 1.11).

The complete migration, or eventual substitution, will take many more years than *experts* first thought. At the end of the day the installed PSTN base is massive, quality is excellent and customer bills are continuously deflating (see Figure 1.12). Incumbent operators should manage VoIP as a complement whilst maintaining the existing POTS and ISDN services. It is significant that a decade after main manufacturers announced VoIP solutions, the traffic penetration still scores below 20% in developed countries, and this level is thanks to early users like youngsters, travelers, and low income users. There are two possible professional VoIP scenarios: a hosted network provided by the operator or private networks based on a VPN or similar.

The challenge of VoIP rollout is not insignificant given the complexity of hybrid solutions that also combine traditional PSTN phone calls with the new packet based solution. Imagine how many protocol translators, gateways and transcodecs that would be necessary to set up a multi-conference between VoIP, POTS and GSM to guarantee compatibility and interconnections.

1.3 Driving factors of Triple Play

Two main groups of driving factors can be identified. Firstly, the necessity to redefine the telecom business and secondly, the consequence of competitive pressure.

1.3.1 Business Redefinition

Telcos initial advantage in voice and data access was quickly blurred by new competition forcing them to operate to thin profit margins and even no profits at all. In order to change the shape of the business, telcos have discovered how a DSL broadband router can be combined with a converged network that would allow the development of a set of new multimedia services.

The strategy starts by offering bundled voice and Internet access paving the way for the introduction of television services to existing subscribers. IPTV will never completely replace broadcast or cable TV, but will complement them and gain significative market share with an interactive platform built around the IP protocol. Video on Demand (VoD), Pay-per-view (PpV), and Video recording services will need to be a bit more innovative.

1.3.1.1 Telcos point of view

Above all, telcos feel that they have finally found, in Triple Play, a unique opportunity to redefine the whole business. This strategy requires a high capital expenditure (CAPEX) to develop a converged network, but will allow innovative product creation by combining legacy and new products and will also reduce operational expenses (OPEX) in an integrated management environment.

At the end of the day telcos expect to increase revenues by means of the new services while keeping existing subscribers loyal to the old ones.

1.3.1.2 Consumers point of view

There is much evidence to suggest that customers are willing to buy a telecom service package, whenever they can get significant savings on their bill for the full service. Customers would also like to reduce the increasing level of complexity in managing their technological devices. Just how many devices, remote controls, configurations, interconnections, can there be in modern homes equipped with PCs, satellite, Hi/Fi, Digital TV, mobiles, Video, Gaming stations, and Surveillance devices?

Residential customers are demanding simple, easy-to-use integrated technological devices that offer more automatic features.

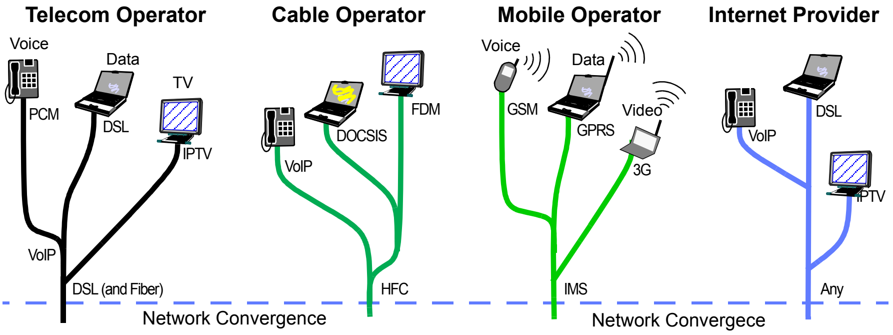


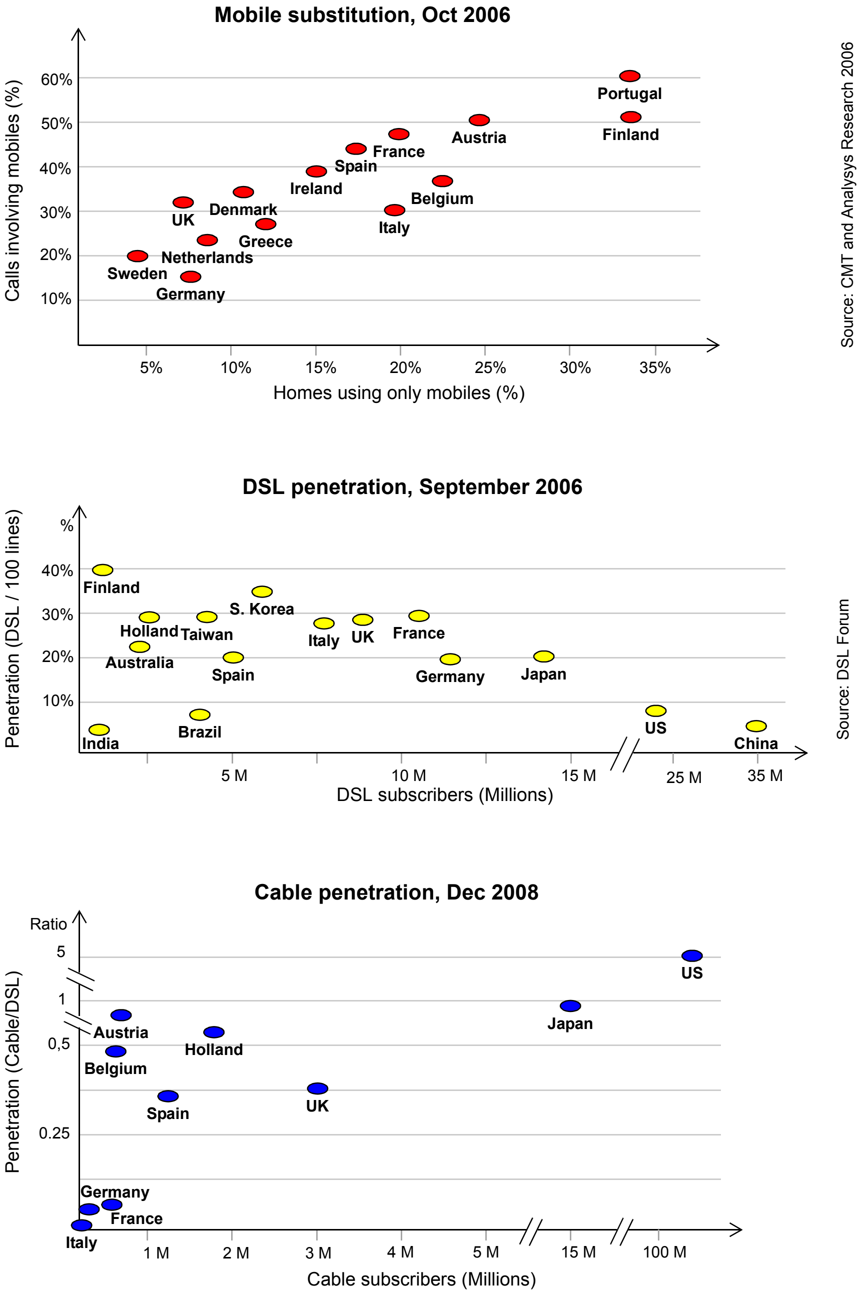
Figure 1.13 Each operator has their own migration strategy for network convergence.

1.3.2 Competitive Pressure

Just a few years ago, companies such as cable operators, ISPs and telcos were different businesses and were not competing with each other. Cable operators were focused on video and TV services, ISPs did not go much further than offering IP data services while the main business of telcos was based on voice and datacom services such as T1, E1, FR, or DSL.

The advance in technology and new regulations have made ISPs, Cable and Mobile operators competitors of telcos in voice and data access. So companies, that were originally in different markets, are all now racing to bundle and offer the same services:

- **Mobile operators**, that are offering wireless technologies have produced a significant social change. Now telephony is a personal service that combines privacy with mobility. It is not a secret that wireless has stolen an important part



Mobile substitution, Oct 2006

Source: CMT and Analysys Research 2006

DSL penetration, September 2006

Source: DSL Forum

Cable penetration, Dec 2008

Figure 1.14 Mobile, DSL and Cable penetrations.

of fixed line revenues, moreover wireless-only operators are constantly suggesting that all telecom services can be delivered using only wireless technologies. Mobile operators, that have invested in 3G infrastructures, are very keen to bring new applications based on voice, Internet, video, and messaging into service. However, it is not very clear what the volume of new revenues will be as multimedia service expectations are not very optimistic (see Figure 1.14).

- **Cable Operators.** New regulations have allowed cable operators to grab the traditional telcos market. Cable operators, that started out offering just TV, were the first to offer Triple Play thanks to the Data Over Cable Systems Interface Specification (DOCSIS). This technology enabled them to also deliver broadband Internet access to their subscribers which later on opened the door to include VoIP as well. (see Figure 1.14).
- **Internet Service Providers**. ISPs probably are the most important threat to the traditional phone service as inexpensive voice calls can potentially reduce voice revenues to a minimum. ISPs are experts on IP services and do not have to manage large infrastructure assets. This fact has enabled them to build a flexible and innovative model around the Internet. VoIP solutions based on software such as Skype are an excellent example of how ISPs are exploiting this opportunity (see Figure 1.14).

The result is that companies, originally in different markets, are now all racing to bundle multiple services using their own version of a converged network (see Figure 1.13). The consequence is that fixed telephony use and revenues are declining after the adoption of mobiles and the penetration of cable operators that also include broadband access and voice services.

Bundling has become a protective strategy for incumbent operators, while in the hands of a competitive operator it is seen as an offensive strategy (see Figure 1.25).

Faced with keen competition from free digital terrestrial television and from cable and satellite providers, telcos have opted for a third way, somewhere between free-to-view and Pay-to-view TV. The difference will be that broadcasters and pay-to-view TV providers will offer a number of premium channels, while telcos will offer a competitive video-on-demand (VoD) library.

1.4 TELCOS STRATEGIES

Strategies for Triple Play depend on a number of parameters such as business size, market position, competition, and existing infrastructures (see Table 1.3). In the case of incumbent operators it is very important to obtain new revenues, while, at the same time, keeping the core and the access business in good shape because this has always been a competitive advantage.

Table 1.3
How to face the competition and how to take decisions depending on business type.

		Competitor						
		Fixed	*3G*	*Quadruple Play*	*ISP*	*Cable*	*Triple Play*	*Satellite*
Strategy	*Fixed*	Price Quality Brand Name	Enrich service VoIP, video-calls, Multiplay. Virtual mobile operator.	Enrich service with VoIP, video calls, Triple Play	Include VoIP Triple Play	Include IPTV, bundle with mobile	Lower Cost, Move to Triple Play	TV, Rural and isolated regions
	3G	Everything on wireless, Move to 3G, HSDPA	Price Quality Brand Name	Lower price, larger pipe	GPRS, UMTS	Mobility, mobile TV	Multi services over GPRS, 3G and HSDPA.	TV, LEO constellation
	Quadruple Play	Convergence of fixed + mobile	Convergence of fixed + mobile	Price Quality Brand Name	Fixed/mobile integration Divert calls to VoIP fixed using Wi-Fi	Fixed/mobile integration	Differentiate with mobile: Quadruple play	TV channels
	ISP	Low cost VoIP calls	Low cost VoIP calls	Low cost VoIP calls	Price Quality Brand Name	Low cost VoIP calls	Low cost VoIP calls	TV channels
	Cable	Include voice, and broadband	Bundle all services enrich customization	Offer HD TV, bundle all services	TV Channels	Contents Price Brand Name	TV Quality and quantity of channels	Contents
	Triple Play	Multiservice simplify billing, enrich contents	Enrich contents	Multiservice simplify billing, enrich contents	Multiservice simplify billing, enrich contents	Focus on contents VoD	Contents Price Support Brand Name	Multiple and simultaneous channels VoD

A well planned strategy should follow a sequence of steps that will pave the way to Triple Play adoption:

1. Broadband access to the fixed line must be periodically upgraded: ADSL, ADSL2+, VDSL2, and FTTP.
2. Progressive bundling of local/national/international calls to line rental and adding the data service to this package.
3. Converged network rollout, to minimize cost in support and maintenance.
4. Mobile/Fixed commercial bundling.
5. Add new applications such as IPTV, VoIP, Video conferencing.
6. Multimedia/Multi-terminal services (see Figure 1.15)

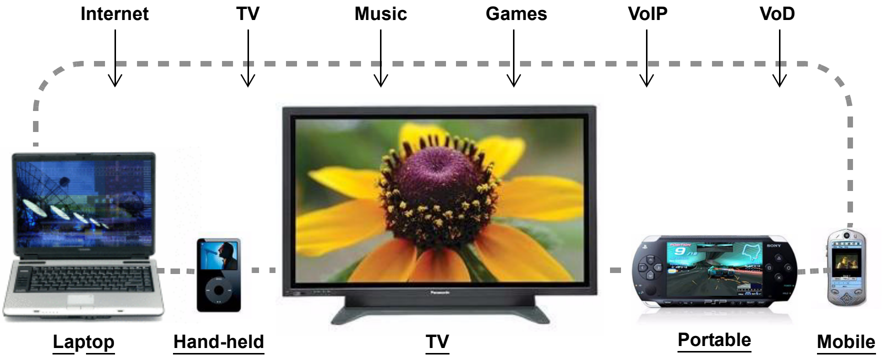


Figure 1.15 Triple Play is not only a matter of multiple information flows but is also about how a wide range of terminals can manage data, audio and video applications.

1.4.1 Service Bundling and Network Convergence

The bundling process starts as a commercial action unifying a flat rate bill consisting of line rental, DSL service and phone calls. Obviously the first step is the integration of local and short distance calls while the most expensive international calls and calls to mobiles should be last, thus reducing the cost progressively.

Service bundling does not require either technological innovations or network migration. It can be accomplished just by means of market engineering and business alliances where necessary.

On the other hand, the network convergence term refers to the network that is able to support all the Triple Play services simultaneously. It also refers to an IP centric architecture that is able to support different service quality accurately.

At the same time, the existing optical transmission core network has to be migrated into a packet friendly core layer to facilitate the interconnection of the metropolitan networks. These networks are being deployed based on bridged Ethernet architectures with QoS enablers like VPLS/MPLS.

Regarding the local loops, fiber will be progressively incorporated using FTTx architectures. Fiber will need to be installed in between the customer premises and the central office to reduce the copper span.

Although, Triple Play strategies may start only with service bundling, migration to a IP centric converged network needs to be part of the involved operators' strategy in order to reduce the delivery costs and simplify the management structure (see Figure 1.16).

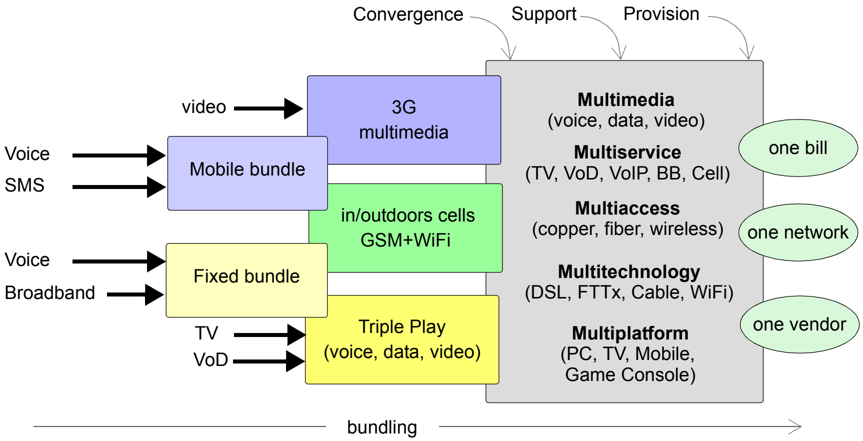

Convergence
Support
Provision
video
3G multimedia
Voice
SMS
Mobile bundle
in/outdoors cells GSM+WiFi
Voice
Broadband
Fixed bundle
TV
VoD
Triple Play (voice, data, video)
Multimedia (voice, data, video)
Multiservice (TV, VoD, VoIP, BB, Cell)
Multiaccess (copper, fiber, wireless)
Multitechnology (DSL, FTTx, Cable, WiFi)
Multiplatform (PC, TV, Mobile, Game Console)
one bill
one network
one vendor
bundling

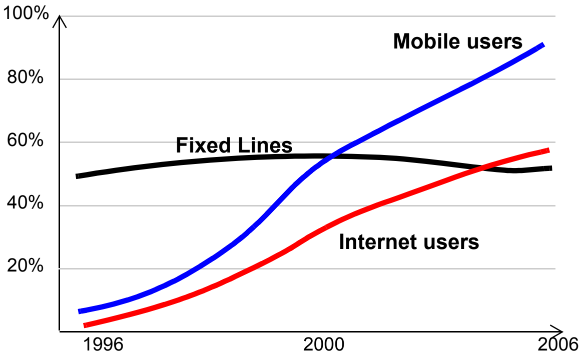
Figure 1.16 Triple Play migration. The Big Game.

1.4.2 Quadruple Play

When cellular phones are added the bundle is often called Quadruple Play. However the new bundle is, not necessarily, a technological convergence, it could be a simple commercial package that only includes bill unification. But it can also be a sophisticated convergence that allows cellular hand-sets to connect home routers to the Internet or to redirect calls through the cheaper fixed line. This convergence happens when the mobile is in the wireless hotspot range (wi-fi or bluetooth) of the router. Obviously, mobiles with wi-fi capacity are necessary to enable such a level of integration by making a seamless switch from the outdoor cell to the router range.

It is interesting to note that mobiles have their own Triple Play strategy, therefore Quadruple play does not really refer to new applications but to the inclusion of the wireless media and cellular hand-sets in the bundle.

Mobile manufacturers and operators have demonstrated that multiple services and technologies can be merged successfully. We find it is quite normal to manage, not only phone calls but also, text and multimedia messages, agendas navigation, Internet access, e-mail, radio, video, gaming, photographs, etc., using only our mobile terminal. Wireless operators have proved that it is possible to maintain or even increase revenues in a saturated market by offering innovative services and applications. These are interesting lessons to be learned by the new bundlers namely, the fixed line operators.



Source: International Telecommunications Union

Figure 1.17 Fixed line, Mobile and Internet penetration rates in developed countries.

1.4.2.1 Financial Integration

Integral operators, with fixed and mobile divisions, have more possibilities to protect their revenues from single network competitors. This strategy puts more elements into the operator's hands so they can offer or, be more aggressive on price than specialized operators. Integral operators can compete much better because they can move forces in both fields simultaneously to complement each other, thus reducing cost and using common staff, and assets. France Telecom and Telefonica are two examples that, by reabsorbing their mobile divisions, have become integral operators.

1.4.2.2 Combined Packages

Combined mobile and fixed services are a natural step forward for integral operators. This can be enhanced with unified features such as call forwarding, unified voice mail, and a common customer service center. This action reduces mobile churn since it is economically convenient for subscribers but it is also more difficult for them to break the bundle or to change all fixed, DSL and mobiles services simultaneously to move to an alternative provider or competitor.

1.4.2.3 Mobile redirection

Calls originated in a cellular network can be redirected to the owner's fixed lines. To achieve this integration the router, and the hand-set, have to establish a wi-fi or bluetooth link when they are close enough. Mobiles can also use broadband access to upload/download data, videos, music or to establish low cost VoIP calls over the Internet. This offer would cannibalize some of the revenues, but would also increase the loyalty of existing customers. Such is the case in Unik FT/Orange and BT Fusion which are bundled services for mobiles that offer low call rates when the mobile user is at home where mobile phones use GSM outdoors and WiFi indoors.

1.4.3 VoD: The Key Difference

IP television is the direct way to face competition from cable operators in their core business. Television is a challenge in many aspects, because it requires a new network, a large access pipe, and rich contents to fill up the channels, moreover TV services is a defensive maneuver that initially will not generate any profits at all.

VoD can reuse an IPTV network, nevertheless it is a different service that requires new protocols, network elements and terminals to be supported. VoD requires rich libraries of specialized contents like films, documentaries, cartoons, sports, etc.

VoD is a key player in the Triple Play offer, since broadcasters (cable operators, satellite, digital) have technical difficulties in implementing this service, therefore it can be seen as the front line to gain new customers and to increase the ARPU of existing.

1.4.4 Making a Success Story

In business there are many paths to failure and only a few to building a success story. Telcos should prepare the path carefully, while the steps will depend on many parameters, the following being some of the most obvious. These are generic trends that every company can customize to their home markets.

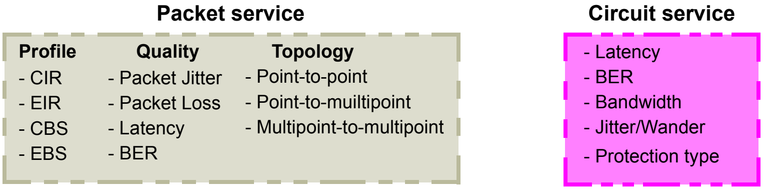
- **Focus on urban customers.** A higher concentration of homes, buildings, and a shorter distance to the central office is convenient for a quick rollout of the access network.
- **Center on high speed connections.** Customers already with broadband access are more likely to contract new bundled services.
- **Cost is the key factor.** Residential customers are very sensitive to cost particularly when contracting commodities such as telephony, TV and broadband access.

- **We also have TV,** telcos have to see TV services as a purely defensive blueprint to keep cable operators under control.
- **Only we have Video,** video services should be the front line strategy since only telcos have the right architecture to support it.
- **Mobile Convergence.** The mobile vs. fixed time is over. Integration of both worlds is strategic.
- **Keep it Simple.** One bill, one provider, is probably less important than a service that is: reliable, simple to manage by the customer and easy to maintain by the operator.

1.5 INFRASTRUCTURES

Operators must keep in mind that the control and management of underlying resources are essential. For a successful Triple Play business case it will be necessary to rollout a converged network that is IP centric and QoS enabled. Independent of the access technology (xDSL, FTTx, Cable, Wi-Max, or Wi-Fi) and the core network architecture, (VPLS, MPLS, Ethernet, NG SDH, WDM, etc.) the converged network must guarantee the QoS to support data, voice or video streams.

Triple Play has different QoS requirements in terms of bandwidth and delays according to the application. For example Video conferencing is sensitive to delay and jitter, but non real-time multimedia is less sensitive to delay and, because it makes use of error recovery techniques, it is less sensitive to packet loss. While data applications are not sensitive to delay and jitter, packet loss may be a critical factor (see Figure 1.18).



Packet service

Profile	Quality	Topology
- CIR	- Packet Jitter	- Point-to-point
- EIR	- Packet Loss	- Point-to-muiltipoint
- CBS	- Latency	- Multipoint-to-multipoint
- EBS	- BER	

Circuit service

- Latency
- BER
- Bandwidth
- Jitter/Wander
- Protection type

Figure 1.18 Parameters to consider when defining a Packet or a Circuit service.

MPLS/VPLS is the mechanism most commonly used to control the QoS in IP networks that are extended to Metropolitan areas.

1.5.1 CPE equipment

Customer equipment varies depending on the access technology and manufacturers preferences but, independent of the access technology (copper, fiber, wireless), we need a router which generally has 10/100BASE-T interfaces, and eventually WiFi, to connect PCs, the set-top-box (STB), and IP phones (see Figure 1.15).

1.5.2 The first mile

During the last decade a lot of progress has been achieved in access technologies. How many are suitable for delivering Triple Play? There are many certainly, including xDSL, Cable, Ethernet, Wireless, and Fiber, the cost to bandwidth ratio being the common selection criteria (see Figure 1.21).

Some telcos are installing Fiber To The Premises (FTTP) which is the best of the possible alternatives available in terms of bandwidth, but the high cost has persuaded many others to continue extracting more bandwidth from their copper pair loops, particularly if they are suitable for the installation of ADSL2+ and VDSL2 (see Figure 1.19). In general Fiber technologies, and specifically the Passive Optical Network (PON), have an associated higher CAPEX because they require fiber deployment to the customer's site; a lower OPEX because all the elements are optical and passive being exactly the opposite of copper technologies.

Cable operators use hybrid fiber coaxial (HFC) a technology that combines fiber optic and coaxial cable for the last span to the customer premises.

1.5.2.1 Digital Subscriber Loop

Copper loops are available everywhere whilst other media must be newly deployed. That is a competitive advantage for the owners of this infrastructure, who are generally incumbent operators. Several DSL technologies are possible, ADSL2+ and VDSL2 being the main alternatives for residential customers. They must guarantee a minimum of 6Mbit/s to support one TV channel (see Figure 1.20).

Unfortunately a high percentage of loops cannot reach the minimum bandwidth requirements for Triple Play. Factors related to physical characteristics and electromagnetic interferences like loop length, bridged taps, crosstalk and attenuation limit the available bandwidth. Attenuation and frequency increases the length of the line. All of these factors are influenced, direct or indirectly, by the distance between the DSL modem, or router, and the DSLAM. This the reason why there is an increasing tendency to reduce the span installing DSLAM in street cabinets, in the neighborhood or in the building.

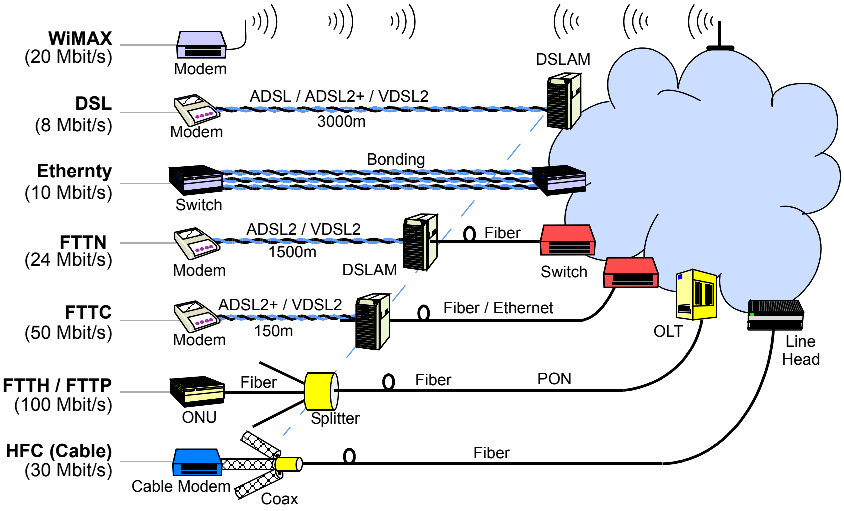


Figure 1.19 Broadband access technologies. Triple Play supports bandwidth hungry applications that require bandwidth of many Mbit/s.

1.5.2.2 Fiber To The Neighborhood

The combination of copper and fiber in the loop is a compromise to offer higher bandwidth at reasonable costs. It is a step forward to continue extracting more bandwidth from the copper pair. Several architectures are possible (see Figure 1.19):

- *Fiber to the Node (FTTN)*, the copper loop part, from the customer home, can be up to 1500 m long to reach the DSLAM which is, in turn, linked to the local exchange with fiber. FTTN has a reasonable rollout[2] cost, while also increasing the number of enabled loops to transport television channels.
- *Fiber to the Curb (FTTC)*, the fiber arrives at a street cabinet that connects homes within a distance of less than 150 m. A higher bandwidth would allow the delivery of more TV channels simultaneously while the crosstalk, attenuation, and noise on the copper are minimized with a shorter wire.
- *Fiber to the Building (FTTB)*, the fiber arrives at the customer's home connecting the CPE and DSLAM over a distance less than 30 m. Higher bandwidth should be achieved because of a shorter copper section.

[2] Per-connection cost in 2006 could be between 300 and 500€.

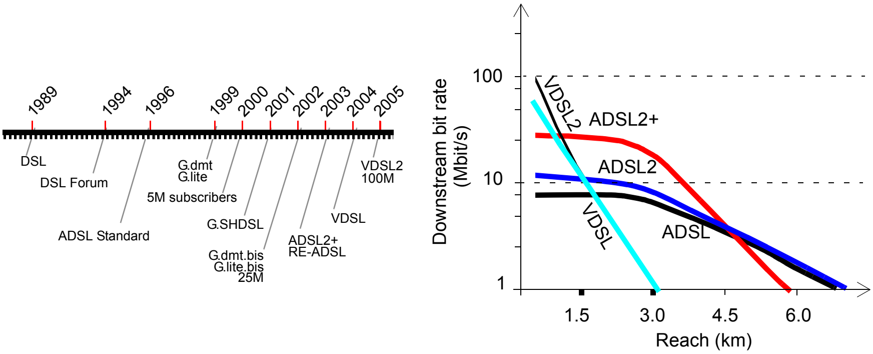


Figure 1.20 DSL has been a real success story in the number of subscribers thanks to its continuous evolution in speed and cost.

1.5.2.3 Optical access

Fiber is the best media to deliver bandwidth higher than 50 Mbit/s but it is necessary to figure out how to make it pay. The capacity of optical transmission is the ideal solution to bypass the bandwidth limitation on the last mile, but it is the most expensive. FTTP, is based on PON or Active Ethernet and can deliver around 100 Mbit/s to each subscriber.

Fiber optic to the home is often a long term strategy particularly appropriate for green-field installations but difficult to justify when alternative access solutions are already installed. According to several consultancy companies a home fiber connection can cost a provider anywhere between €600 and €2000 per subscriber depending on how many difficulties arise. Existing neighborhoods are the most expensive because it is often necessary to dig a ditch to reach every single home. In new developments it is easier and much cheaper when fiber is installed together with water, gas, electricity and other utilities.

Source: DSL Forum

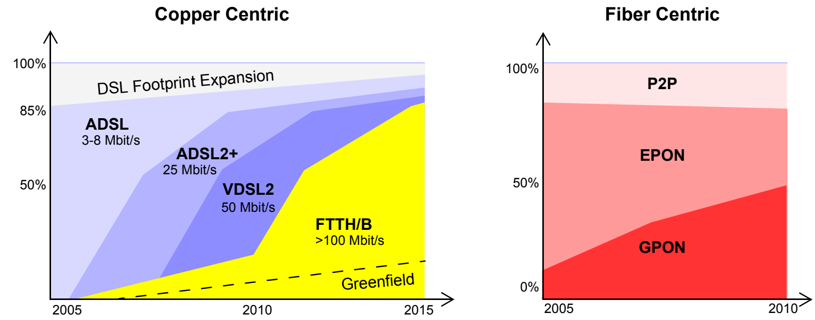


Figure 1.21 Forecast of broadband access technologies.

1.5.2.4 Cable access

Cable operators use Hybrid Fiber Coaxial (HFC) to deliver dozens of simultaneous TV channels thanks to the big bandwidth pipe this technology can support. DOCSIS architectures enabled Internet access and, later on, VoIP was possible due to a very aggressive strategy to increase the market share of residential consumers of telecom services. However, telcos have used xDSL2+ as the answer to deliver TV and VoD. This has made the cable operators business much more difficult.

1.5.2.5 Regulations

Regulatory factors may modify all the access strategies depending on how the unbundling legislation, that was applied to the copper local loops, affects the new access infrastructures:

- Will the regulations for unbundling the local loop be canceled?
- Will the cable/fiber network be unbundled? If so it will depend on the unbundled resources: voice frequencies, packet traffic or full capacity.
- Will Fiber to the Network (FTTN) together with wired access, be unbundled?

Obviously owners of fixed lines are applying pressure to prevent the introduction of more regulations that would allow third parties to use new optical and hybrid infrastructures.

There is a significant difference between what is happening on both sides of the Atlantic regarding regulations. In the US recent decisions are backing companies that are investing in infrastructures, while in Europe regulations affecting fiber optics are similar to those affecting the copper local loop.

It is also significant that the US has suppressed some of the unbundling obligations of the owners of cable infrastructures. According to the FCC the intention is to stimulate the investment in new FTTx infrastructures, then FTTN access will be offered at a reasonable cost to competitors but not at a regulated price. This decision is supported by the fact that alternative operators already have had time to develop their own business and access networks.

In Europe new directives will abolish all regulatory distinctions between networks, telephone, and the Internet, by placing all services and networks into a single, all-encompassing, regulatory category called ‘electronic communications’. The consequence is that some European countries have opened up sub-loop access and optical access as well. These differences explain why the European Commission and National Governments are under pressure from the dominant operators to adopt the FCC model and keep new access technologies unregulated. The incentive is to justify the massive investments that will be required. The result is heterogeneous, while in Germany BNetzA has permitted Deutche Telecom to close its VDSL-FTTN to competitors, while in the UK the regulator OFCOM has opened up sub-loop access.

1.5.3 Network Convergence

Despite the widespread availability of nx64 circuits, IP and Ethernet, two packet oriented technologies, have been selected to build the Triple Play network (see Figure 1.18). The statistical multiplexing, typical of packet technologies, has important advantages. The first is cost, as they are much cheaper to rollout and maintain compared with circuit networks based on SDH. The second: IP and Ethernet are easier to manage and have a lot of synergy with the Internet which is based on the same principals. But probably the ability to implement any new service based on voice, data and video is even more important than the benefits mentioned above.

1.5.3.1 Is the IP Protocol Ready to Do it?

IP is considered the best strategy to adopt in the deployment of the new converged services and, particularly television, which has for a long time been offered exclusively by cable, terrestrial and satellite operators. The IP protocol differentiates telco portfolios from competitors. Native IP-centric infrastructures that were developed for data transport, must be transformed into a multiservice platform that is also able to transport audio and video. To achieve this important investments are required.

The TCP/IP protocol, was designed in 1983 and immediately adopted by the US Department of Defense to connect heterogeneous hosts and has demonstrated itself to be very robust and suitable for managing large and complex topologies. It is also Internet-like and requires a minimum of human intervention. The Internet that we know, developed during the 1990s, is an architecture that provides universal connectivity between heterogeneous but open subsystems. TCP/IP protocols are designed to automatically locate topologies and addresses by means of nodes that are continuously interchanging routing information.

IP networks are robust enough to maintain data flows between hosts but do not necessarily use the shortest or the most efficient route. It means that packet delay is unexpected and can even vary during the time a session is active. In IP networks what really matters is keeping datagrams flowing from source to destination whilst ensuring that they are independent of any events that can affect individual nodes. They are, in some respects, fault tolerant by nature.

Legacy TCP/IP architectures are best effort solutions that are good enough to transport data but are inappropriate for supporting Voice or Television which demand, not only an accurate QoS control, but also 99.999% availability, high performance and protection mechanisms. To achieve this the layered TCP/IP suite has

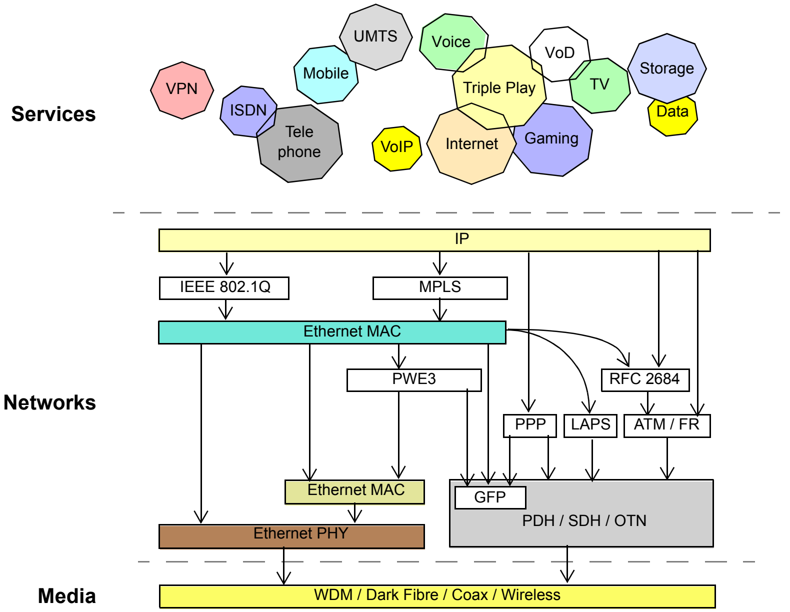


Figure 1.22 Applications, Services and Protocols

adopted a number of protocols to emulate the predictable but inefficient circuits based on SDH. The result can be even better and more robust than legacy services when the right architecture is adopted (see Figure 1.22). Elements like SIP, RTP, RTCP, VPLS, or GFP are only some of the protocols that can help IP to implement carrier grade networks to support Isochronic applications based on data but, also on voice and video/TV.

1.5.3.2 Ethernet at Layer 2

Ethernet has a number of features that have made it the favorite technology for implementing architectures to support Triple Play services. Ethernet scores high in a combination of features like efficiency, simplicity, scalability, and cost. It is also important that Ethernet is the technology used in the vast majority of customer premises and service provider installations.

Ethernet is efficient as it is packet oriented, therefore it obtains the statistical multiplexing gain when transporting independent traffic flows over shared transmission medias. Ethernet is also very simple to set up and maintain especially when compared with SDH-SONET installations. Other important considerations are the number of engineers and technicians, probably millions, that are confident with Ethernet and all its associated devices and protocols such as switches and TCP/IP.

IEEE 802.3 Ethernet versions. List of acronyms: H/F: Half-Duplex and Full-Duplex ability. MFS: Minimum Frame Size in bytes. N/A: Not applicable. MMF: Multimode Fiber. SMF: Single Mode Fiber. U: Up Stream. D: Down Stream

					Encoding			
	Interface	**Media Type**	**Mbps**	**FDX**	**Data**	**Symbol**	**MFS**	**Reach**
Ethernet IEEE 802.3a-t	10BASE-2	One 50 Ohm thin coaxial cable	10	H	4B/5B	Manchester	64	<185 m
	10BASE-5	One 50 Ohm thick coaxial cable	10	H	4B/5B	Manchester	64	<500 m
	10Broad36	One 75 Ohm coaxial (CATV)	10	H	4B/5B	Manchester	64	<3600 m
	10BASE-T	Two pairs of UTP 3 (or better)	10	H/F	4B/5B	Manchester	64	<100 m
	10BASE-FP	Two optical MMF passive hub	10	H	4B/5B	Manchester	64	<1000 m
	10BASE-FL	Two optical MMF asyn hub	10	F	4B/5B	Manchester	64	2000 m
	10BASE-FB	Two optical MMF sync hub	10	H	4B/5B	Manchester	64	<2000 m
Fast Ethernet IEEE 802.3u	100BASE-T4	Four pairs of UTP 3 (or better)	100	H	8B/6T	MLT3	64	<100 m
	100BASE-T2	Two pairs of UTP 3 (or better)	100	H/F	PAM5x5	PAM5	64	<100 m
	100BASE-TX	Two pairs of UTP 5 (or better)	100	H/F	4B/5B	MLT3	64	<100 m
	100BASE-TX	Two pairs of STP cables	100	F	4B/5B	MLT3	64	200 m
	100BASE-FX	Two optical MMF	100	F	4B/5B	NRZI	64	2 km
	100BASE-FX	Two optical SMF	100	F	4B/5B	NRZI	64	40 km
Gigabit Ethernet IEEE 802.3z/ab	1000BASE-CX	Two pairs 150 Ohm STP (twinax)	1000	F	8B/10B	NRZ	416	25 m
	1000BASE-T	Four pair UTP 5 (or better)	1000	H/F	4D-PAM5	PAM5	520	<100 m
	1000BASE-SX	Two MMF, 850 nm	1000	F	8B/10B	NRZ	416	500/750 m
	1000BASE-SX	Two MMF, 850 nm	1000	F	8B/10B	NRZ	416	220/400 m
	1000BASE-LX	Two MMF, 1310 nm	1000	F	8B/10B	NRZ	416	< 2 km
	1000BASE-LX	Two MMF, 1310 nm	1000	F	8B/10B	NRZ	416	<1 km
	1000BASE-LX	Two SMF,1310 nm	1000	F	8B/10B	NRZ	416	5 km
	1000BASE-ZX	Two SMF, 1310 nm	1000	F	8B/10B	NRZ	416	80 km
10GEthernet IEEE 802.3ae	10GBASE-SR	Two MMF, 850 nm	10000	F	64B/66B	NRZ	N/A	2 to 300 m
	10GBASE-SW	Two MMF, 850 nm	10000	F	64B/66B	NRZ	N/A	2 to 33 m
	10GBASE-LX4	Two MMF, 4 x DWM signal	10000	F	8B/10B	NRZ	N/A	300 m
	10GBASE-LX4	Two MMF, 4 x DWM signal	10000	F	8B/10B	NRZ	N/A	300 m
	10GBASE-LX4	Two SMF, 1310 nm, 4 x DWM	10000	F	8B/10B	NRZ	N/A	10 km
	10GBASE-LR	Two SMF, 1310 nm	10000	F	64B/66B	NRZ	N/A	10 km
	10GBASE-LW	Two SMF, 1310 nm	10000	F	64B/66B	NRZ	N/A	10 km
	10GBASE-ER	Two SMF, 1550 nm	10000	F	64B/66B	NRZ	N/A	2 to 40 km
	10GBASE-EW	Two SMF, 1550 nm	10000	F	64B/66B	NRZ	N/A	2 to 40 km
Ethernet First Mile IEEE 802.3ah	100BASE-LX10	Two SMF, 1310 nm	100	F	4B/5B	NRZI	N/A	10 km
	100BASE-BX10	One SMF, 1310 (U)/1550 (D)	100	F	4B/5B	NRZI	N/A	10 km
	1000BASE-LX10	Two SMF, 1310 nm	1000	F	8B/10B	NRZ	N/A	10 km
	1000BASE-LX10	Two MMF, 850 nm	1000	F	8B/10B	NRZ	N/A	550 m
	1000BASE-BX10	One SMF 1310 nm (U)/1490 nm (D)	1000	F	8B/10B	NRZ	N/A	10 km
	1000BASE-PX10	One SMF, PON 1310 nm (U)/1490 nm (D)	1000	F	8B/10B	NRZ	N/A	10 km
	1000BASE-PX20	One SMF, PON 1310 nm (U)/1490 nm (D)	1000	F	8B/10B	NRZ	N/A	20 km
	10PASS-TS	One or more telephone pairs	10	F	64/65-octet	T1.424	N/A	750 m
	2BASE-TL	One or more telephone pairs	2	F	64/65-octet	G.991.2	N/A	2.7 km

Ethernet is designed to be used with many types of optical and metallic media including the most popular. Transmission ranges and bandwidths are equivalent to long haul technologies. Being easily scalable from a few Mbit/s up to many Gbit/s (see Table 1.4) it is therefore possible to migrate existing LANs, MANs and WANs to Ethernet using the existing physical media

1.5.3.3 Ethernet drawbacks

Unfortunately, native Ethernet lacks some essential functions necessary to supply 'carrier-class' services. Features like reliability, management, rollout, maintenance and QoS are much more demanding in Metro networks supporting Triple Play than LANs where Ethernet is focused on data transport. Scalability can also be an issue. Ethernet switches work very well when the number of hosts connected is limited and during low traffic conditions, but as soon as the installation grows it tends to degrade in performance, QoS, security, availability.

When Ethernet is extended beyond the LAN, several architectures can fulfil the requirements including Dark Fiber, DWDM / CWDM, NG-SDH (see Figure 1.23).

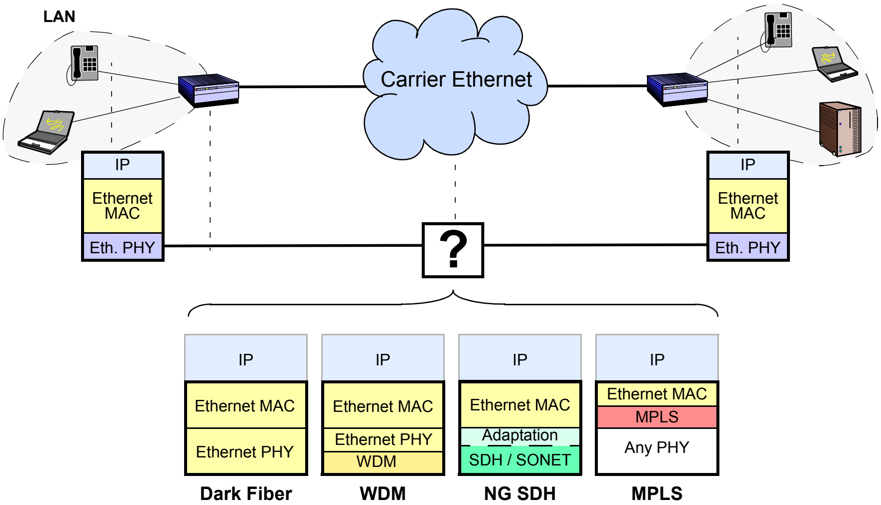


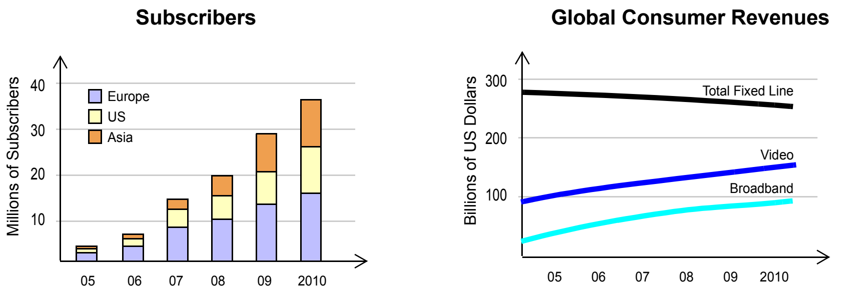
Figure 1.23 Alternatives for providing Carrier Ethernet services.

In principle, all of these architectures are able to support the Carrier Ethernet services like E-Line and E-LAN, nevertheless some are more appropriate than others.

When deciding on which architecture to use most of these native Ethernet limitations are passed over. Depending on the requirements of the applications; budget; customer profile; installed base; whether there is any optical infrastructure, and if so, how much capacity does it have; are more important factors to consider.

1.6 Triple Play market

Important growth that can be expected in the residential market for Triple Play services in the US has been calculated at 32 million annually (Yankee Group Aug. 2006), with an average operator spending rate of about $4,000 per subscriber. Pyramid research is not so optimistic, calculating a world market of 35 Millions by 2010 (see Figure 1.24). According to Gartner research, the current average combined monthly spend that includes fixed voice, Internet and TV in Europe is €93,70. This roughly matches Fastweb reports in Italy of obtaining an ARPU of €900 a year so that the ARPU can be increased by 100% after the adoption of bundled services (Heavy Reading, 2006)


Subscribers

Global Consumer Revenues

Source: YGR

Source: Yankee Group Research, Inc. Sept 2006.

Figure 1.24 Triple Play subscribers and revenues prediction.

Forrester Research warns that low revenue opportunities and the high investments needed for broadband Triple Play could result in significant losses for telecoms operators. Forrester (www.forrester.com) predicted (April 2006) an initial cumulative loss higher than €3,000 per subscriber.

Caution is necessary. Cable companies such as NTL in the UK and FastWeb in Italy which could be considered as Triple Play providers, are not growing quickly, and in some cases, are even losing market share. Put quite simply: many potential subscribers are not interested in having one provider for the three services.[3]

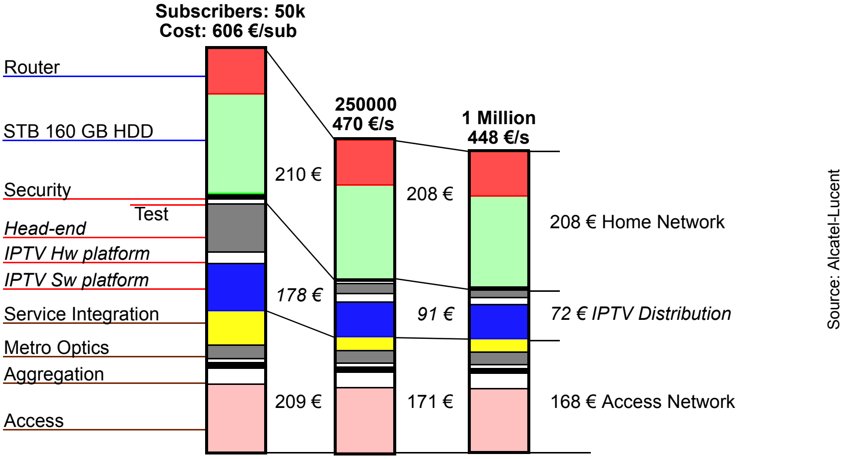
[3] Source: Les Echos 26/Sep/2006 and Others.

1.6.1 Warning: no immediate profit is expected

1.6.1.1 Providers view

The size of the broadcast and cable TV market has been calculated to be about 25% of the whole telecom business, and this is a good reason, for telcos to go into this business. But the provision of TV has a major impact on telecommunications architectures. Massive investments are required to transform legacy networks into multiservice platforms.

This is the million dollar question. Will service providers get any benefits at all? Triple Play's main message is 'more services for less money'. How is it really possible for operators to increase revenues? If the answer is not clear for Triple Play operators it is even more difficult for single service operators. How can they compete against an efficient multiservice network that can deliver any type of application and also integrates mobiles?



Source: Alcatel-Lucent

Figure 1.25 Triple Play Deployment Cost and the scale economies.

1.6.1.2 Customers view

The growing enthusiasm for bundling video with existing services demands high investments but customers are not willing to pay a lot for new TV channels since it is a mature service already supplied by broadcast, cable, digital, terrestrial TV, and satellite (see Figure 1.10).

Therefore, from a subscriber point of view, the true value of Triple Play is not exactly the addition of TV; it is the convenience of a bundled package that can be simply managed and cheaper bills.

Competition is now even more extreme but the customer base remains the same with the services being presented in a different format.

Therefore it is urgent to stimulate the demand with new services and applications that can generate more revenue. Otherwise many companies could be at risk of not recovering their investments.

1.7 Conclusions

Services bundling and network convergence are different facets of the same strategy, since it is not possible to offer a competitive package, with many varied services, without a network that reduces delivery costs and simplified management.

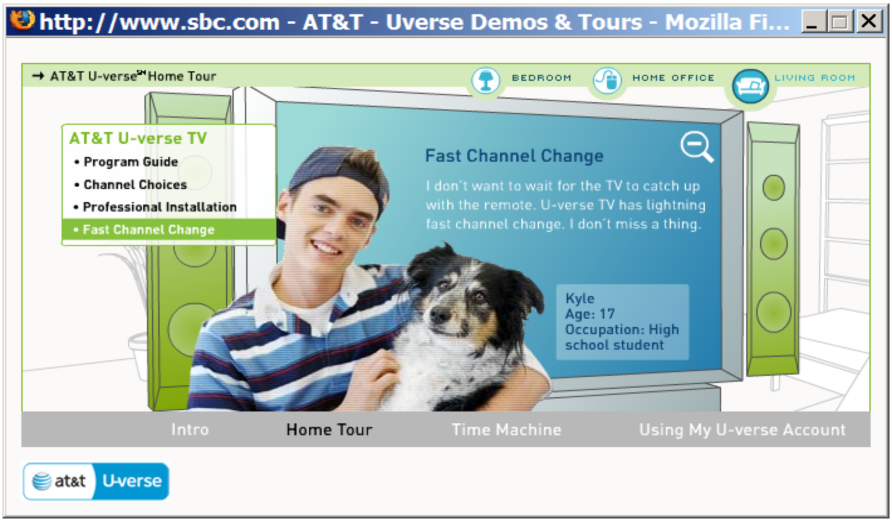
Table 1.5
PTV Customer offering[i]

Service Provider	*Movies*	*TV Contents*	*Songs*	*Other*
T-Com Germany	1,500 titles	1,500 hours	Unknown	Sport Services
Chunghwa Telecom	Unknown	Unknown	6,000	Karaoke
Telefonica	200 titles	1,500 hours	1,600	Sport Services
Virgin Media	500+ titles	Last week TV	1000+	Adult Services
Sky TV	200+ titles	Up to 40 hours	Unknown	Sport Services
BT-Vision	200+ titles	Unknown	150+	15+ Games

i. Source: Alcatel-Lucent

Bundling is the strategy to adopt in order to compete and differentiate services in mature telecoms markets. Service providers with mobile and fixed networks are moving towards a wholesale business by integrating separate divisions. Single service operators are signing up to alliances with other operators to enrich their services to offer a more comprehensive portfolio that would ideally, include wireless, fixed access, voice, video and data services. This is also the case for some fixed line operators that also offer wireless access or those mobile operators that have opened up their network to offer wire based services.

The convergence of technologies and services is a trend that most operators are having to face (see Figure 1.26). But, will this strategy produce profits? The question is how to bundle more services for less money while combining both mobile and fixed networks.



http://www.sbc.com - AT&T - Uverse Demos & Tours - Mozilla Fi...

Fast Channel Change

I don't want to wait for the TV to catch up with the remote. U-verse TV has lightning fast channel change. I don't miss a thing.

Kyle
Age: 17
Occupation: High school student

Figure 1.26 Fading phone business pushes companies in a new direction: Triple Play, like AT&T and its U-verse service offers: IPTV, VoD, broadband access (wired and wireless), music downloads, gaming, integrated video recording, integrated ISP portals, VoIP, IP video conferencing and file sharing.

It is still too early to say if Triple Play services will be commercially successful. It is a new technology with many technical challenges and scalability problems ahead. Cable operators are not taking any risks at the moment because they already are one step ahead as they have already added broadband access and voice to television, their core business. But we should not underestimate the greater variety of services that telcos can offer with the flexible IP-based architectures.

Regarding service rollout, there are a number of technical challenges to fulfill. Voice, television, video and broadband access all have different traffic profiles and place different requirements on the network that delivers these services. Voice and television are affected by jitter, while packet loss has a greater effect on Video and Data services. Consequently accurate Quality of Service mechanisms are necessary to develop and control the delivery of Triple Play.

In some ways we can find certain similarities with old convergence strategies that happened in the industry during the last few decades. Some of them were successful, like the ISDN convergence of voice and data, but others for example ATM, were not at all able to streal the limelight on the convergence stage. Another recent case of a failed convergence has been 3G, when wireless European operators paid a fortune to operate 3G networks that would also deliver a bundle of voice, data and video services. It must be frustrating when the new 3G services are sold as just offering low cost calls.

Therefore companies embracing bundling and convergence should take precautions to avoid repeating the same errors.

Chapter 2

IP Telephony

Telephony is the technique that enables transmission of human speech over long distances with the help of electromagnetism.

The first days of telephony were dominated by analog signal transmission and Frequency Division Multiplexing (FDM) of multiple voice signals. Digitalization was a major advance in telephony. By means of digitalization, voice signals, that are continuous by nature, are represented by discrete sequences of symbols belonging to a finite alphabet. Digital signals have many advantages: they can be stored or replicated without limit, they are resistant against noise and they can be efficiently encrypted, compressed and filtered by Digital Signal Processors (DSPs).

The Pulse Code Modulation (PCM) can be used to represent a voice signal by a sequence of ones and zeros. This fact is very important because historically, PCM was the first modulation used in digital telephony. PCM signals are well suited to Time Division Multiplexing (TDM). The first digital telephony systems, based on PCM and TDM, were introduced in the sixties.

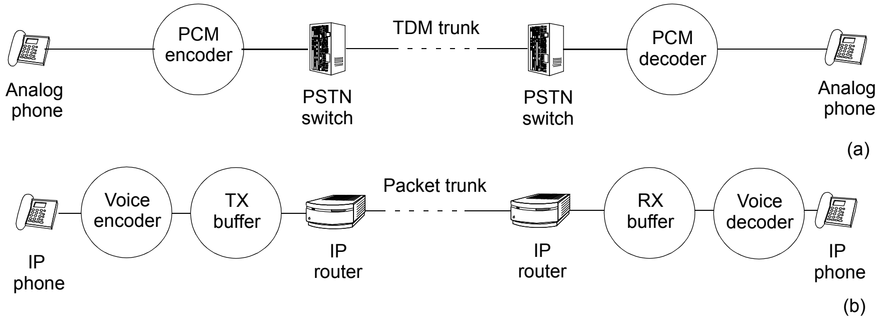


Figure 2.1 Reference models for telephony services. (a) classical telephony reference model. An analog signal is generated by a telephone, digitalized with a PCM encoder and transported in TDM trunks. (b) IP telephony reference model. Voice is encoded, digitalized, assembled in packets and transported in packet trunks. Buffering is necessary to deal with variable delay.

Currently, the world of telephony is facing a new revolution: Deterministic TDM is being replaced by Statistical TDM which is based on packet switching. The high availability and low cost of IP equipment makes this technology a popular alternative for transporting voice when compared with ATM or FR. Reliability and quality of IP telephony, or Voice over IP (VoIP), have improved over the last few years and today it is a fast growing industry (see Figure 2.1).

IP telephony is a new way to provide a service that is more than one hundred years old. From this point of view, IP telephony alone does not bring anything new to subscribers. They would not even notice migration to IP transport infrastructures. However, IP telephony can potentially bring benefits to subscribers, service providers and carriers:

- Subscribers benefit from lower call costs because operators can avoid paying interconnection charges. IP telephony gives new service providers a competitive edge over incumbent operators because it gives them more and more flexible deployment options for their services. It enables new service providers to avoid many of the regulatory barriers of standard voice provision. Of course, it is thought that subscribers will also benefit from this scenario of increased competition.
- For carriers, IP telephony brings the ability to integrate voice and data over the same network infrastructure. This makes installation and management tasks easier and cheaper.
- It is accepted that VoIP will cut the benefits that service providers will get from telephony but, on the other hand, IP telephony offers new opportunities by means of a new class of services that go beyond classical telephony such as videotelephony, integration of mobile and fixed telephony or telelearning. It is expected that many of the business opportunities for service providers will come from the possibility of bundling classical and innovative services together.

From the technical point of view, the main advantage of IP telephony is that it enables the convergence of data and voice. But convergence is not possible without QoS aware IP networks. During the last few years a lot of effort has been concentrated on designing the architecture of such networks (see Paragraph 7.2). However, most of today's IP networks, including the Internet remain best-effort networks.

Given the current state of the art of IP technology, integrating time-sensitive and other traffic into Internet backbones is not necessarily a good idea if the carrier is to offer VoIP with quality approaching that of PSTN. Deployment of a dedicated IP network to carry voice is the option chosen by some operators. For them, integration of voice and data is left to a second stage, once the network has been prepared to deal with different traffic classes.

Both integrated and dedicated VoIP networks suffer from QoS problems that were not present in classical telephony. Due to queuing effects, delay in IP networks is higher and far less predictable than in Deterministic TDM networks. To correct the effects of variable delay, some traffic has to be buffered by the receiver before sending it to the decoder. This buffer, is in fact a dejittering filter that must never become full or empty during the telephone call to avoid losing data or leaving the decoder without voice samples. The need for a dejittering buffer increases the overall delay experienced by the voice samples. Larger buffers favour jitter absorption but they increase the end-to-end delay. This trade off between delay and jitter filtering capabilities of a VoIP receiver is an important fact in IP telephony. For non interactive applications such as IP radio there is no compromise and dejittering buffers can be larger without any noticeable effect experienced by the end user.

2.1 Coding of Voice Signals

Transmission of digital voice signals involves at least two steps:

1. *Sampling*: The analog voice signal obtained from a microphone is converted into a sequence of discrete samples.
2. *Coding*: Every sample is assigned to a symbol taken from a finite alphabet. Usually, symbols are bit words.

The result from the previous two operations is a bit stream that can be transmitted by a digital channel. To be efficient, the sampling rate of the voice sampler must be as low as possible without compromising the quality of the stream. Something similar happens with the encoding process. The average number of bits generated for every sample must be as small as possible.

Voice sampling and coding take advantage of how the sound is perceived by our auditory organs and how voice is generated. Another point to be taken into account is the required quality level of voice for telephony applications. Telephony users want a clear and easily understandable voice. They will probably value the ability to understand the message and identify the speaker but other features such as high fidelity or stereophonic sound would probably be unnecessary.

Human hearing is able to perceive sounds between about 20 Hz and 20 khz, but most telephony standards only require the transmission of frequencies between 300 Hz and 3400 Hz. While this is only an small sub-band within the band of all audible sounds, it contains about 80 % of the sound information from original voice signals. For transmission systems, adapted for music transmission like FM radio, the whole audible band is required to be transmitted, but for telephony, between 300 Hz and 3400 Hz is considered enough.

Standards for legacy digital telephony systems specify PCM coding of voice signals. PCM is simple and well suited to TDM transmission but things are different in IP technology. Packet transmission can take advantage of variable bit rate codecs. An example of voice codec for packet transmission is one that only generates data when users are speaking but it does not generate information during silent intervals. The available range of voice codecs for IP telephony is broader than for TDM telephony. Voice codecs for packet transmission can be classified into three different families:

1. *Waveform codecs* encode the voice signal without taking into account how the signal was generated. They try to quantify the voice samples in such a way that the signal decoded by the receiver resembles the original signal as much as possible. They produce high quality output with low coding delay, while maintaining low complexity, but they are not well suited to low bit rate transmission. Another advantage of these types of codec is that they are useful for the transmission of non voice signals such as Dual-Tone MultiFrequency (DTMF) digits.
2. *Source codecs* or *vocoders* attempt to emulate the way speech is produced. The parameters of the model are sent to the receiver instead of the waveform itself. Voice is modeled by an excitation passing through a linear predictive filter that models the vocal tract. Human speech sounds can be voiced like an "a" or unvoiced like an "s". The way these sounds are generated is fundamentally different. This is the reason why two different excitations are needed. Vocoders break down the voice signal into short chunks of a few milliseconds and then find the parameters of the model that include a flag to choose the excitation type, the parameters of the excitation and the coefficients of the linear filter for each chunk This usually involves fewer bits than waveform transmission. However, voice synthesized by the receiver sounds non natural and it is difficult to identify the speaker and therefore vocoders are hardly ever used for telephony applications. The Linear Predictive Coding (LPC) is the most famous source codec.
3. *Hybrid codecs* attempt to provide the best of waveform and source codecs. They try to match the waveform of the original signal but at the same time they use models based on how speech is generated by humans. Hybrid codecs are complex and they usually produce a meaningful coding delay. Bit rates obtained from them are a cross between vocoders and waveform codecs but the performance often approaches that of source codecs.

The availability of many voice codecs has the advantage that service providers and subscribers are free to choose the best for their needs. In VoIP applications, more than one codec may be used in the same end-to-end connection since cellular, broadband access and PSTN networks usually require different codecs. These transcoding operations degrade the voice quality. Analysis of distortion caused by transcoding is complex and the penalty paid for the end quality may be appreciable.

2.1.1 Pulse Code Modulation

Pulse Code Modulation (PCM) is a waveform coding technique that assigns every sample from the signal to a number represented as a binary word. The PCM coding is specified in ITU-T Recommendation G.711 for transporting digital telephony signals in the PSTN.

The chosen sampling frequency for voice applications has been always 8 kHz, slightly more than twice the maximum frequency contained in telephony voice signals (3400 Hz). This is in line with what is required by the Nyquist theorem that states that an analog signal of bandwidth B sampled with frequency of at least $2B$ can be recovered without error from the sampled version. On the other hand, every sample of the voice signal is represented by an 8 bit word as specified in G.711. This gives a transmission rate of 64 kb/s.

The 64 kbit/s is the base for all TDM multiplexing hierarchies and 125 μs, the inverse of the sampling frequency, is the duration of all common TDM frames.

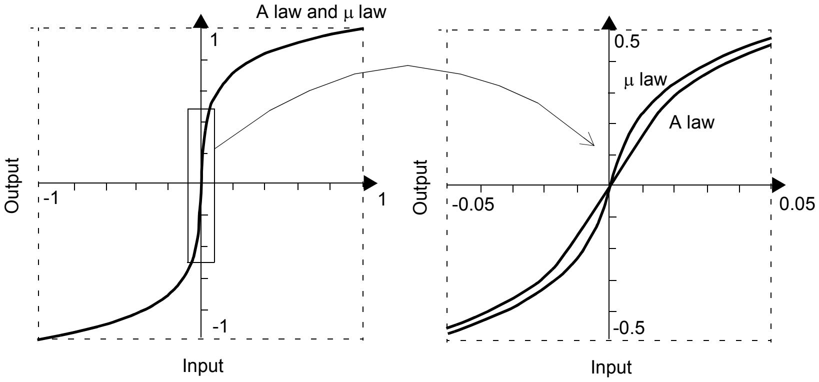


Figure 2.2 Comparison of the A law and the μ law. The comparison curves are nearly identical.

The amplitude of a voice sample is a real number but there are only 256 possible 8 bit words. That means that quantification involves an error that is perceived by the receiver as random noise. The quantifier is chosen to minimize this error. Specifically, G.711 quantifiers are not uniform. In the US and Japan the quantifier follows the μ law in most other countries it follows the A law. The shape of the μ and A laws are nearly identical and none of them are uniform (see Figure 2.2).

Statistical analysis shows that smaller amplitude values of voice samples are more likely to occur than larger values. The reason is because the shape of the μ and A law quantifiers are designed to give more accurate codifications of samples with small amplitudes. Another reason is that human hearing is more sensitive to variations in weak sounds and it is therefore practical to minimize quantification error in these samples.

2.1.2 Adaptive Differential PCM

The Adaptive Differential PCM (ADPCM) coding is an evolved version of the basic Differential PCM (DPCM). Both ADPCM and DPCM are examples of waveform codecs. With DPCM encoding it is quantified and delivered as the difference between two consecutive samples rather than individual voice samples as in PCM.

DPCM works because voice signals are not completely random. There is some correlation between samples taken close in time. When computed, the difference between two consecutive samples increases the chance of obtaining a value that is smaller in absolute value than the individual samples. The difference can be quantified with fewer bits than in the PCM case and still maintain good performance. At the receiving end, the previously decoded sample and the received DPCM sample, can be used to compute the current sample. Without transmission errors the original and the received sequence must be the same.

The ADPCM is a sophisticated version of the DPCM and is based on the concept of linear prediction. So instead, the difference between the current PCM sample and the previous sample is computed and the difference between the current sample and an estimation of its value is obtained from previous samples. DPCM is a particular case of ADPCM with an estimation based only on the previous sample. ADPCM works in the same way that DPCM does. Correlation between voice samples enables estimation. The encoder is said to be adaptive because the coefficients of the linear estimation change with time to match the statistics of the signal to be transmitted.

In statistical terms, the estimation obtained from ADPCM is better than from DPCM and the differential sample can be coded with fewer bits. The receiver must implement the same adaptive linear estimator in order to be able to decode the differential samples.

The ITU-T Recommendation G.721 describes a standard ADPCM codec to deliver voice at 32 kb/s, one half of the PCM rate. The G.721 has been superseded by Recommendation G.726 that encodes voice signals at 16, 24, 32 or 40 kb/s.

2.1.3 Code-Excited Linear Predictive

Code-Excited Linear Predictive (CELP) codecs are examples of hybrid codecs. Specifically, CELP codecs are examples of Analysis by Synthesis (AbS) codecs, the most important family of hybrid codecs. They are based on the same modeling principles as LPC. Both have a linear predictive filter to model vocal tract but, models for AbS codecs are more complicated than the model for vocoders because they involve more than two excitations. AbS codecs have a collection of excitations for the linear predictive filter. The encoder finds the output of the filter for all the possible excitations and then computes the distance between the original and the synthetic signal. Finally it sends the coefficients of the filter and a pointer to the excitation that constitutes a better approach to the original signal, to the receiver.

There are many implementations and variations of the CELP codec. Some of them are included in standards for their use in IP telephony applications.

Low-Delay CELP (LD-CELP) has been standarized in ITU-T Recommendation G.728. It works with 5 samples simultaneously that are sampled at 8 kHz. It finds optimum filter coefficients and excitation from a codebook that contains 1024 items. LD-CELP receivers are able to use previously received information to compute the current filter coefficients and therefore, all that is necessary is to send is a pointer to the excitation signal. This information can be coded in 10 bits since there are 1024 excitations. The resulting bit rate is 16 kbit/s with good voice quality compared with for example, 16 kbit/s ADPCM. LD-CELP introduces a delay of 5 samples. Less than 1 ms delay is not noticed by the human ear.

Another codec based on a variation of CELP is called Conjugate Structure CELP (CS-CELP) and is standarized in ITU-T Recommendation G.729. The CS-CELP algorithm works with blocks of 80 samples sampled at 8 kHz. For these blocks the codec computes optimum excitation, filter coefficients and gain parameters and assembles 80 bit frames that are sent to the destination. This results in a bit rate of 8 kbit/s. The coding delay of the G.729 codec is 15 ms.

Recommendation G.729 contains various annexes that further enhances the performance of the basic codec adaptation operation to specific environments (see Table 2.1).

Table 2.1
Extensions to the basic G.729 codec

Standard	*Purpose*
G.729A	Simplified codec compatible with G.729 but with slightly worse performance.
G.729B	Algorithm specifications for Voice Activity Detection (VAD), Discontinuous Transmission (DTX) and Comfort Noise Generation (CNG). These algorithms are used to reduce the transmission rate during silence periods of speech.
G.729D	Lower bit rate (6.4 kb/s) version of the basic algorithm for environments with reduced available bandwidth.
G.729E	Increased bit rate (11.8 kb/s) version of the basic codec specially adapted for environments with background music at the input.

2.1.4 Other Codecs

Although the codecs discussed above are the most common and important, they are just a small selection of the available alternatives (see Table 2.2). An important subset of voice codecs are those defined for digital mobile communications. Mappings into RTP exist for most of these codecs. This make codecs for wireless voice available for VoIP applications as well.

Table 2.2
Other Voice Codecs

Codec	*Rates*	*Main Applications*
A-CELP (G.723.1)	5.3 kb/s	VoIP
MP-MLQ (G.723.1)	6.3 kb/s	VoIP
CDMA Q-CELP	6.2 kb/s, 13.3 kb/s, others	IS-95 CDMA wireless systems
GSM FR	13 kb/s	GSM wireless systems
GSM EFR	12.2 kb/s	GSM wireless systems
AMR	from 4.75 to 12.2 kb/s	GSM and 3G wireless systems

2.2 Network Performance Parameters

Quantities such as Signal to Noise Ratio (SNR), Bit Error Ratio (BER) or end-to-end delay give information about network performance. However, it is difficult to infer what the users' perception of the network will be when based only on these ratios. The converged network offers heterogeneous services to the subscribers. The perception users have of the network depends on the service but, SNR, BER or delay are service agnostic.

Service agnostic performance parameters are still useful but their relationship must be investigated together with the perception that users receive from the service. In other words, the values of performance parameters must be interpreted for a given service.

For voice, two main QoS parameters are important: packet loss and one-way delay. Delay variation is less important because it can be filtered by the receiver at the price of increased one-way delay. However, delay variation may potentially cause buffer overload or underload at the receiver when the buffering mechanisms are not well dimensioned.

2.2.1 Packet Loss and VoIP

Packet loss (see Paragraph 6.2.3) causes distortion in voice applications that increases as the packet loss ratio increases. When the overall quality of the voice service is represented in terms of the Mean Opinion Score (MOS) (see Paragraph 2.3) the MOS is seen to decrease. (see Figure 2.3).

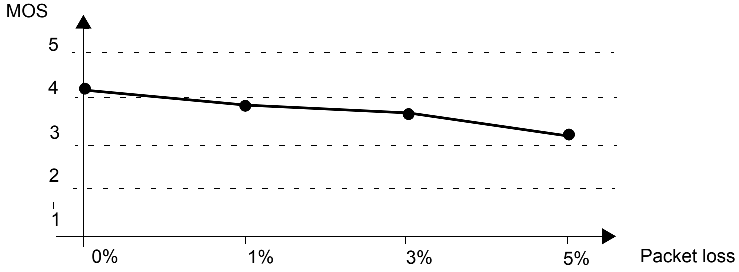


Figure 2.3 Performance degradation of the voice signal expressed in terms of the MOS as a function of packet loss. The considered coding is PCM (G.711 Appendix I) and packet length is 20 ms.

The ability to keep the quality high with occasional lost packets, depends on the following points:

- *The distribution of the lost packets*. Uniform distribution of lost packets tends to be easily absorbed by the decoders. Bursty distributions cause deeper loss of performance.

- *The Packet Loss Concealment (PLC) algorithm in use*. PLC algorithms provide packets for the decoder when the actual packets are lost. A PLC may insert silence, background noise, repeat the last packet received or use interpolation techniques to replace lost packets.

In general terms, packet loss has minimum impact on voice quality if the packet loss ratio is less than 1 %. A good PLC helps by keeping quality for loss ratios to about 3 %. Higher loss ratios cause severe degradation to the perceived quality. Another factor to take into account is the effect of packet loss depending on the codec. PCM codecs exhibit more tolerance to lost packets than more complex voice compression algorithms. Specifically the G.711 codec is more tolerant to errors than the G.729 and G.723.1 codecs.

2.2.2 Delay and VoIP

As one-way, end-to-end delay increases in the telephony application, switching between the roles of talker and listener becomes more difficult. This undesired effect degrades the quality of all types of interactive communications and must therefore be controlled. Dealing with delay is one of the biggest challenges of IP telephony.

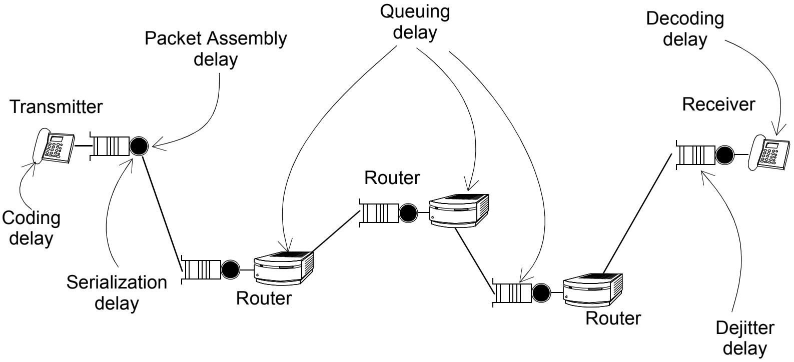


Figure 2.4 Delay sources in the VoIP service.

The delay experienced in IP telephony used to be higher than for traditional telephony. The reason is that delay is increased by new elements not present in circuit switched networks (see Figure 2.4). These are the more meaningful sources of delay for VoIP:

- *Coding / Decoding delay.* Sophisticated codecs achieve high compression of the speech data but pay the price of delay. There are two different contributions to the coding delay: the processing delay and the lookahead delay. The former is the time it takes to process a voice frame, the latter is the time the decoder waits for future data to exploit correlation with the frame currently being processed, to achieve better compression. We have to add to the coding delay, the decoding delay; which used to be about one half of the coding delay. The overall coding / decoding delay can be as little as a few tenths of a millisecond. One of the important voice encoders, the G.729 CS-CELP, has a coding delay of 15 ms of which 10, are processing delays and 5 are lookahead delays. For the same algorithm, the typical decoding delay is 7.5 ms.

- *Packet Assembly delay.* Many voice encoders generate very short frames. The G.711 encoder generates an 8 bit frame every 125 μs and the G.729 generates an 80 bit frame every 10 ms. In order to increase the transmission efficiency the transmitter may choose to fill every IP packet with various voice frames. This improves the ratio of user versus overhead bytes but results in increased delay before transmission. For example, the packet assembly delay to generate an IP packet with 5 G.729 frames would be 50 ms.
- *Serialization delay.* This is the time it takes to place one packet on the transmission line. It depends on the size of the packet to be transmitted and the transmission rate in the access link (see Paragraph 6.2.1). For a 100 byte frame and a 64 kb/s line, the serialization delay would be 12.5 ms.
- *Queuing delay.* End-to-end delay in the network is known as increased queuing delay (see Paragraph 6.2.1). Queuing delay appears when the network is experiencing congestion. Queuing delay is variable and difficult to predict but it can be minimized by implementing QoS mechanisms in the network. Other delays that happen within the network like switching / routing delay or propagation delay are usually not as critical as the queuing delay.
- *Dejittering delay.* To allow variable packet arrival times and still achieve a steady stream of packets, the receiver holds a certain amount of data before playing them out into a buffer. This is the dejittering buffer and the time that packets wait in the buffer is known as the dejittering delay. The dejittering delay has to be added to the overall end-to-end delay in the VoIP application. Normal sizes for the dejittering buffer are about 50 ms but this depends on the variability of packet delay. Many commercial solutions offer intelligent buffers that automatically adjust according to the network performance.

The ITU-T Recommendation G.114 advises that the overall end-to-end delay for voice communications can accumulate up to 150 ms without being noticed by the communicating parties. When delay exceeds 150 ms negative effects gradually appear. After calculating all of the possible delay contributions, it seems that achieving the performance level stated in G.114 is not trivial in IP telephony.

The 150 ms delay limit given by G.114 is simple, clear and enables the design of IP telephony systems. However, it has been seen in practice that is difficult to establish a clear limit to acceptable delay in telephony systems. Some authors advise that 200 ms is still a good figure and some others confirm that up to 400 ms can be accepted in long distance calls. An important reason that makes it difficult to agree on the delay limit, is that the degradation effects of delay depend on a new parameter: the echo.

In traditional telephony, the main source of echo are imperfections caused by hybrids. Hybrids are circuits that adapt four-wire systems used within the telephone and two-wire systems used in the local loop. Hybrids are not needed when all the communicating parties are directly connected to an IP network. In other cases hybrids are still used and may cause echo in VoIP communications but even without hybrids echo may appear caused either by the coupling between the speaker and microphone in the telephone or crosstalk.

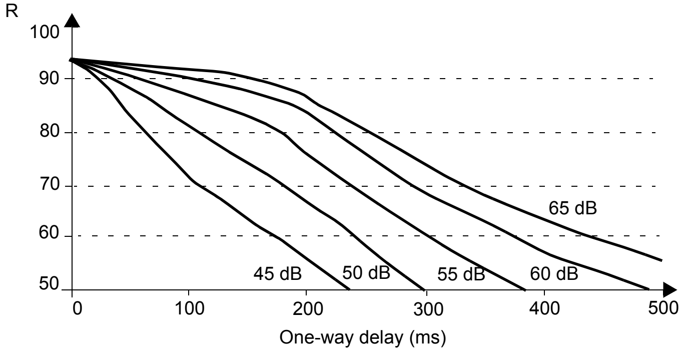


Figure 2.5 Incidence of echo on voice quality expressed in terms of the R factor that measures the quality of the service on a scale of 100 points. The numbers next to the curves represent the Talker Echo Loudness Rating (TELR) measured as the overall two-way attenuation of the voice signal, measured on the path between the transmitter and the receiver.

It has been seen that echo and delay cause interdependent perception of degradation in voice applications. This interdependence can be summarized as follows: echo sounds louder to speakers as delay between original signal and replication increases. In other words, echo may remain unnoticed when delay is controlled but is perceived by the speaker as a degradation when delay is high enough (see Figure 2.5).

A popular solution to echo, both in traditional and VoIP systems, are echo cancelers based on Digital Signal Processors (DSPs) that can reduce the strength of echo by up to 25-30 dB.

2.3 Opinion Quality Rating

Opinion quality rates for voice provide information about how the telephony service is operating by taking opinions from users as an input. These ratings have two objectives:

1. *Qualify voice distortion that is due to encoding, decoding and transcoding*

operations. To measure codec distortion it is necessary to find a test scenario where any other degradation can be neglected. For example, by directly chaining the encoder and the decoder.

2. *Qualify the service globally*. Thanks to this capacity, opinion rates are often considered very useful when rating voice services. The inconvenience is that they are service specific. In other words, service performance values for audio may not be useful for video or data applications.

It is worth noting that none of the above mentioned tasks can be achieved with QoS parameters such as delay or packet loss ratio only.

The basic opinion rate for voice is computed by averaging the subjective perception of users. A group of users is requested to rate the quality of the voice in a telephony service as “excellent”, “good”, “fair”, “poor” and “bad”. The opinion of each user is mapped into a numeric scale (see Table 2.3), and then the arithmetic mean is computed from all the values obtained from the group. The number given by this methodology is called the Mean Opinion Score (MOS) and is an attempt to build an objective parameter from many individual, subjective opinions by means of statistical mechanisms.

Table 2.3
MOS scale

Value	*Meaning*
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

The MOS is defined under ITU-T Recommendation P.800. It is part of a family called Absolute Category Ratings (ACRs) because quality is rated without a reference. ACRs are opposite to Degradation Category Ratings (DCRs) which are based on the comparison between quality perceived by users in the received voice signal and a reference signal. It is possible to compute a parameter similar to the MOS but that takes degradation ratings as inputs instead of absolute ratings. This new parameter is referred to as Degradation MOS (DMOS).

When evaluating the MOS it is useful to distinguish between listening quality and conversational quality. The later involves evaluation of interactivity in a two-way conversation. Encoding and transmission techniques may be designed to minimize voice distortion but they may neglect end-to-end delay. If this happens, the listening MOS will not be affected but conversational MOS will show this degradation.

The MOS is very useful to compare the performance of different voice codecs (see Figure 2.6). Over all codecs commonly found in practice, the G.711 for PCM at 64 kbit/s offers the best score. Other codecs such as the G.728 or G.729 have an score that is only slightly worse than the PCM but they are more efficient in terms of bit rate.

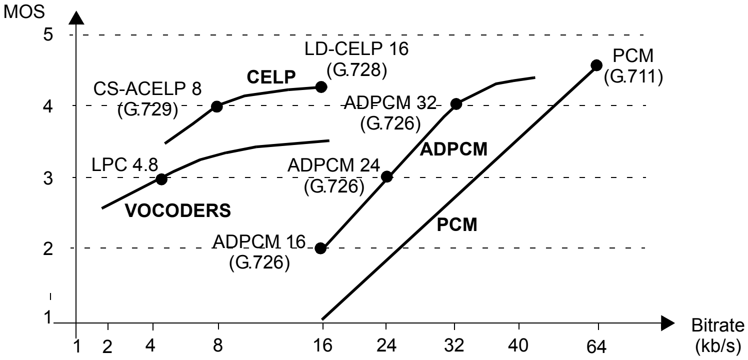


Figure 2.6 Performance of different voice encoders as a function of the bit rate accomplished and the MOS score.

2.4 Objective Quality Assessment

Evaluation of the MOS is complex, time-consuming and expensive. This is the reason why in practice, direct evaluation of the MOS is replaced by estimations performed by machines or algorithms rather than asking people to rate the service. Algorithmic rating methods are said to be objective because they do not rely on opinions but many of them attempt to estimate the MOS, a parameter obtained from averaging individual opinions.

Depending on the input data they take, objective quality assessment algorithms can be classified into the following groups:

- *Opinion models*: These take as inputs a variety of factors, including delay and other network and terminal performance parameters and produce estimates of conversational MOS. The most important and widely used opinion model is the E-model.

- *Speech-layer models*: Require speech signals as inputs and they then generate estimates of listening MOS.
- *Packet-layer models*: In this case the required inputs are IP packets and the output is again an estimate of the listening MOS. Packet-layer models are useful for monitoring speech quality at intermediate network points, where speech signals are not available, but they are necessarily limited because the perceived service performance depends on the network and the terminal features, but the characteristics of the terminal are unknown and from intermediate points.

2.4.1 The E-model

The E-model is a computational model that uses transmission parameters to predict the subjective quality of voice. This model was proposed for standarization in the nineties and it is based on an NTT model called OPINE. Today, the E-model has been standarized in the ITU-T Recommendation G.107. It has also been adopted as a network planing tool by the European Telecommunications Standards Institute (ETSI) and the Telecommunications Industry Association (TIA).

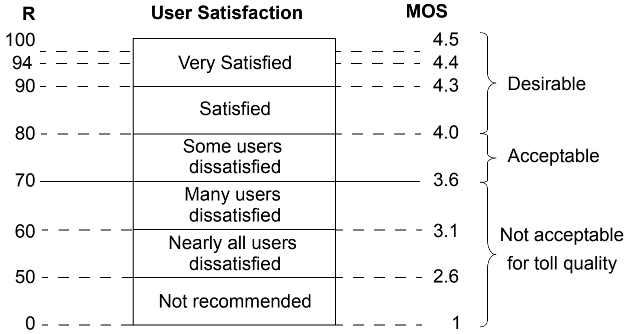


Figure 2.7 Relationship between the R-factor and the MOS, according to G.107 Annex B and G.109.

It provides results in terms of the R-factor that rates the quality of the speech on a scale of between 0 for terrible and 100 for perfect. The R-factor values can be theoretically mapped into the MOS scale (see Figure 2.7). Due to its limitations, 94 is the maximum value for R attainable by conventional PCM voice over circuit switched networks. For VoIP systems, the normal values for the R-factor are usually worse.

A basic assumption of the E-model is that the effects of individual degradations can be calculated to take account of the psychological perception that all these degradations cause to the user. The R-factor is calculated as follows:

$$R = R_0 - I_s - I_d - I_e + A \qquad \textit{eq. 2.1}$$

Where R_0 is based on the Signal to Noise Ratio (SNR), which considers background and electrical noise, I_s captures impairments due to quantizing noise, sidetone and other effects that happen simultaneously with the voice signal generation. The I_d and I_e terms quantify the effects of delay related impairments (including echo) and distortion of voice due to coding and lost packets within the network. Finally, The factor A , called the advantage factor, accounts for the willingness of users to accept substandard performance on a new service.

It has been seen that the E-models consider the effects caused by delay and loss of interactivity in voice communications. This is the reason why this model predicts the conversational MOS rather than the listening MOS.

2.4.2 Speech-layer models

Speech-layer models make estimates of the voice quality perceived by the users taking the voice signal as an input. The first and most basic speech-layer quality evaluation tool is the SNR defined as the ratio between the received voice and noise signal powers. Although simple, the SNR, defined as a ratio of physical quantities, does not consider the psychological effects of impairments. This is the reason why more sophisticated parameters were designed.

The first accurate speech-layer model was the Perceptual Speech Quality Measure (PSQM), which is the basis for the ITU-T Recommendation P.861. When it was standarized the PSQM was tested against other algorithms in order to find out which of them provided maximum correlation with opinion ratings.

The idea behind the PSQM is to measure degradation due to distortion in a “psycho-acoustic domain” rather than in the more conventional time or frequency domains. This involves mapping signals into a domain where, to compute distortion needs proper cognitive modeling and involves a sequence of complex operations such as Fast Fourier Transforms (FFTs), frequency warping, filtering, etc.

The PSQM operates by comparing the degraded and the original signals. That means that PSQM tests are always performed with a reference. The algorithm gives an estimation of the MOS that is sometimes called Objective MOS (OMOS) because it is computed by an algorithm instead of being the result of subjective perceptions. The result from the PSQM algorithm represents the listening MOS rather than the conversational MOS because delay related impairments are not taken into account.

The Perceptual Analysis Measurement System (PAMS) goes beyond PSQM. The PSQM is mainly designed for quality assessment of speech codecs but it is unsuitable for today’s voice networks, especially IP telephony, because it is unable to deal with packet loss and variable delay. These problems were solved by the PAMS algorithm. Currently the importance of PAMS lies in the fact that is a forerunner of the Perceptual Evaluation of Speech Quality (PESQ) algorithm, the latest and most accurate of the speech-layer objective quality assessment models. In fact, the PESQ combines the excellent time alignment algorithm of PAMS and the psycho-acoustic model of PSQM+ (extension of the PSQM).

The PESQ, as it has been standarized in ITU-T Recommendation P.862, follows this sequence to estimate the performance of the voice signal:

1. It corrects the level of the degraded and the reference signals. After this operation both signals have the same average power.
2. Degraded and reference signals are individually processed by a filter that emulates the receiving device (handset, loudspeaker or headphones).
3. The two signals are time aligned to compensate for time shifts.
4. In order to account for the distortions actually perceived by a human listener, degraded and reference signals are transformed into the psycho-acoustic domain by an auditory transformer.
5. Psycho-acoustic representations of both signals are subtracted and the difference is processed by a block that models human hearing (cognitive model).

As a result of this sequence of operations an estimation of perceived voice quality is obtained that can be mapped to the MOS scale.

2.5 Market Segments

The success of IP telephony is favouring the development of commercial solutions specially designed for various market segments from PC to PC voice communications over the Internet, to solutions for service providers intending to replace circuit-switched with packet-switched infrastructures. The main IP telephony market segments are listed below:

- *Single user solutions*: These are the simplest VoIP solutions. They are for users that want to replace or complement costly PSTN calls with low cost VoIP calls. These are often software based solutions integrated into multimedia enabled PCs.
- *Enterprise solutions*: These constitute a market segment directed at companies interested in integrating their voice communications into their IP infrastructure. Most of the currently available solutions for enterprises are based on modified routers with VoIP gateway and Private Branch eXchange (PBX) features.
- *Carrier solutions*: These enable carriers who are providing voice services to a large quantity of residential customers or businesses over a packet based infra-

structure. Current VoIP solutions for carriers are based on large residential and trunk VoIP gateways, sophisticated QoS-enabled routers and intelligent signaling modules.

2.5.1 Single User Solutions

There are popular VoIP software solutions that enable voice communications between remote users across IP networks such as the Internet. Some of these software solutions are based on open standards like H.323 or SIP. Furthermore, some sites in the Internet act as SIP proxies or PSTN gateways and offer presence and other useful services at low cost or even free.

The main advantage of these solutions are cost and simplicity. However they do not provide performance in line with a carrier class service like traditional telephony does. Specifically, today's most popular software based solutions exhibit QoS and availability problems because they use the best-effort Internet as their transport infrastructure.

2.5.2 IP Telephony in Enterprise Networks

IP Telephony products for enterprises exist for companies who want to migrate their infrastructure for voice communications to IP. The new IP infrastructure can be their own Enterprise data network, often based on Ethernet, or WAN data links and circuits provided by an external service provider. Other time advantages of VoIP are limited to external calls while internal calls are still delivered through existing VoIP or ISDN based equipment and cabling.

VoIP enabled IP routers are the main component for some enterprise solutions. One example is the Cisco solution that also uses the CallManager software for signaling and call control of Cisco integrated solutions or third party developments. Another approach to Enterprise VoIP is to upgrade or replace legacy PBXs with VoIP PBXs. The solution based VoIP router is more focused on integral IP deployments without legacy telephony equipment and cabling. The solution based on VoIP PBXs should be considered, if taking advantage of low cost VoIP calls while maintaining existing internal POTS or ISDN infrastructures, is required.

VoIP enabled routers or PBXs must be capable of delivering calls from / to the PSTN, over a Virtual Private Network (VPN) or over the LAN, depending on the destination. For example, IT technicians may choose to deliver internal calls over the existing LAN, calls directed to a remote subsidiary may be delivered to a VPN configured between delegates of the same company and finally, all other voice calls can be routed to the PSTN (see Figure 2.8). The call routing process must be transparent to users. They don't need to be aware of the underlying transport technology for their calls.

A number of issues appear when voice traffic has to share transmission resources with data traffic generated by other applications. Since VoIP applications are sensitive to delay and jitter, voice performance damage may result. On the other hand, VoIP traffic can potentially degrade performance of data applications. Operation bandwidths are 100 Mbit/s or 1000 Mbit/s in most current LANs but VPN links are built over WAN links usually with smaller bandwidths per subscriber. That means that it is likely that VPN links can become a bottleneck for the traffic. To deal with this issue, it would be necessary to implement traffic differentiation, customized treatment of traffic classes, resource management and other QoS mechanisms into the corporate network.

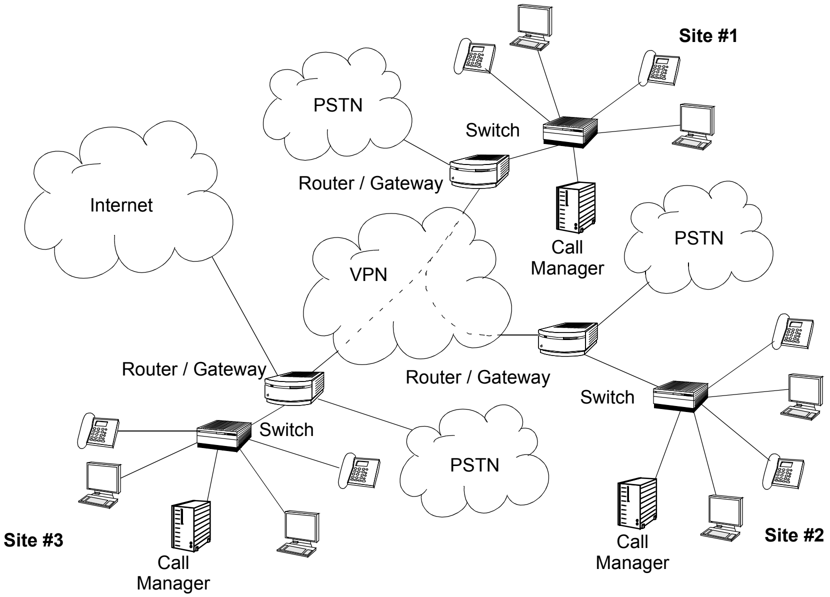


Figure 2.8 VoIP for enterprises. IP telephones are connected to the company IP based data network. Actual transport of voice calls becomes transparent to users. The system is able to route the calls to the best transport network (IP based or PSTN) according to the destination and other established rules.

Another issue, regarding VoIP communications in Enterprise Networks, is security. By default, voice traffic is not encrypted by most of the applications and may potentially be sniffed and can heard by anybody. VPN links are usually encrypted to provide basic confidentiality for data communications and therefore there is no need to perform modifications here, but IT personnel must consider the need to implement security mechanisms in the LAN links in order to provide confidentiality within the local network.

Furthermore, problems arise when the transmission of VoIP calls through a firewall due to limitations of protocols for transporting packetized voice and signaling (see Paragraph 4.3.7), is required. These limitations must be considered and solutions based on STUN, TURN or other mechanisms must be implemented.

2.5.3 IP Telephony in Service Provider Networks

IP Telephony products for carriers are designed to provide voice services to the general public and businesses. There are two important requirements that any carrier class solution for IP telephony must address:

1. *Interworking*: IP telephony has its own signaling protocols based on SIP or other standards. The implications of interworking between VoIP signaling and traditional signaling must be studied for each particular deployment.
2. *QoS*: The performance provided to subscribers must be at least as good as the performance given by traditional telephony. This involves careful planning and dimensioning, deployment of QoS mechanisms in the transport network or even segregation of the packet-switched voice network and the data network if necessary.

The previous issues are complemented by the requirement of a smooth transition from circuit switched networks to the packet switched network. Migration should be as transparent as possible to end users. This is the reason why, perhaps the most important product needed by service providers for their deployments, are gateways between PSTN and IP networks (see Paragraph 4.3.8). Residential gateways integrate voice communications into the transport IP network while maintaining traditional interfaces with subscribers.

For carriers and service providers, monitoring, accounting and other management tasks are critical. Some of these functions exist, and are implemented in the market gateways, but it is possible to run them in separated boxes as well.

Another issue that carriers must keep in mind is a smooth integration of the fixed and mobile networks. Telcos are now integrating their fixed and mobile businesses to ease the provision of bundled mobile and fixed multimedia services which are sometimes referred to as Quadruple Play. But, Quadruple Play involves not only service bundling, but also technological convergence between fixed and mobile networks. In fact, mobile voice communications are going through the same stages as fixed voice. After digitalization, mobile voice communications are now facing packetization. The first solutions are ATM-based but IP is expected to come onto the scene quickly and replace ATM. The all-IP fixed / mobile network has to face different challenges. Among them the:

- Introduction of IP transport technology into the interfaces of the radio access network. In other words, replacement of ATM switching by IP routing.
- Addition of an IP Multimedia Subsystem (IMS), that involves migration of all services and signaling to packet-based infrastructures. This includes voice, video, data and mobility aspects (for example, handover).

Chapter 3

Audiovisual Services

Video and television applications delivered the thought technologies that the Internet made popular are seen by telcos as an alternative business to compensate the declining revenues obtained by traditional voice and data applications. Nevertheless, audiovisual services are, so far, new for telcos since previous experiences were no more than transparent transport of TV signals operated by broadcasters. In consequence telcos are in front of a totally new business case that must be defined to target mainly consumers of entertainment services. Telcos should also learn how to manage contents in order to offer attractive channels that could compete with existing TV operators that include cable, terrestrial and satellite broadcasters.

Audiovisual Services is a big market, the demand exits, therefore it will not be necessary to create it. That is a good news, particularly after the WAP and 3G failures, but the question is, there is room enough for a new type of TV operator? We do not know yet, but it is important to bear in mind three points:

1. First, consumers willingness to watch TV is declining, particularly the new generation is more captivated by the Internet and digital gaming platforms.
2. Second, TV revenues are not so big than it may be supposed, in West Europe the telecom business is about four times the whole television business.
3. Third, and probably the most important, innovative TV applications that combine television and Internet, can create a new market.

IPTV service will be successful only if it is able to differentiate itself from the existing broadcasters in quality and innovative features. Generally talking IPTV strategies should go for a premium service as it can enjoy using so many possibilities to define a new television paradigm binding the audiovisual signals with the internet tools that would permit to go beyond the traditional TV. An IPTV program has many possibilities to be enriched with web sites, video-clips retrieval, promotional items, collaterals, or chats, that ca perfectly be embedded and offered during the program showtime.

New entrants must find the way to differentiate the service, by using the possibilities of the bidirectional features of the IP protocol and the processing capacity of the distribution network. This will be a true IP television (IPTV[1]) a new audiovisual service on subscribers enjoy the ability to control when and where receive or download programs and videos, while combining internet-like interactions.

3.1 Digital Television

There are two basic models to deliver digital TV, the unidirectional used by broadcasters and the bidirectional adopted by IPTV operators:

1. **Unidirectional model**. All TV channels are multiplexed and sent simultaneously over the transmission media until reaching receivers. Subscribers have to tune the program they want to watch. This is the model adopted terrestrial, cable, satellite operators.
2. **Bidirectional model**. Each channel is streamed individually in IP packets to those subscriber that have previously requested it. This model, which relays on the Internet suite of protocols, is more complex to deploy, but it is more flexible, and allows the development of the new interactive applications,

3.1.1 The Internet and Television

IPTV is made up as a flexible combination of two major elements, the IP suite of protocols and digital television:

1. The IP suite of protocols, are responsible of packaging TV signals that will be be routed from the head-end to the customers site. The IP protocol also provides with interactivity features between subscribers, network, service and content providers.
2. The Digital Television, is responsible of the audio, video, data compression and transmission formats according standards, such as MPEG, that manage image resolution and programs arrangements.

Once we have arrived to this point, it is important to clarify that nowadays the public Internet can´t support today real time television services because of several reasons. *First*, the Internet basically is a best effort network that can’t guaranty permanently

[1] When the acronym IPTV is used stand-alone, generally refers to all the applications that deliver audiovisual programs by means of IP packets including both live TV) as well as stored Video on Demand (VOD) and personal video services (PSV). When IPTV is used in a context together with VoD, PSV it would refer exclusively to the multicast in real time television services.

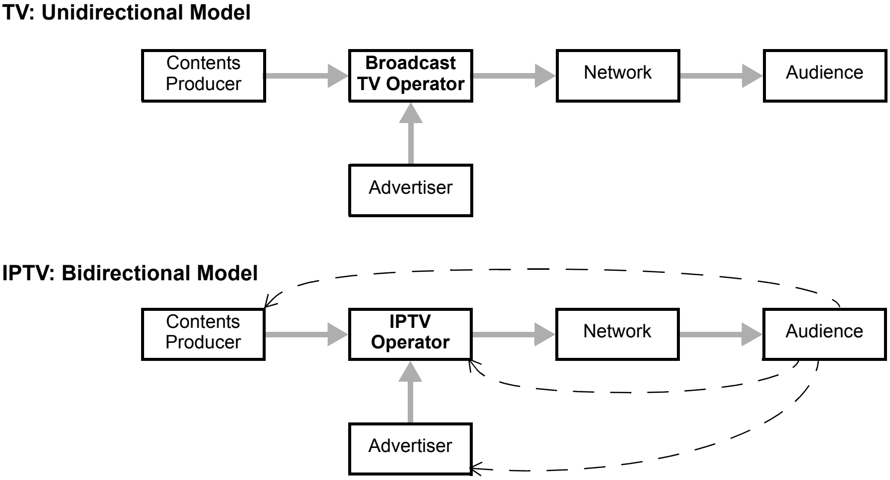


Figure 3.1 In the traditional broadcast TV model there is not direct interaction with the audience, but the IPTV model permits a closer interaction with the audience.

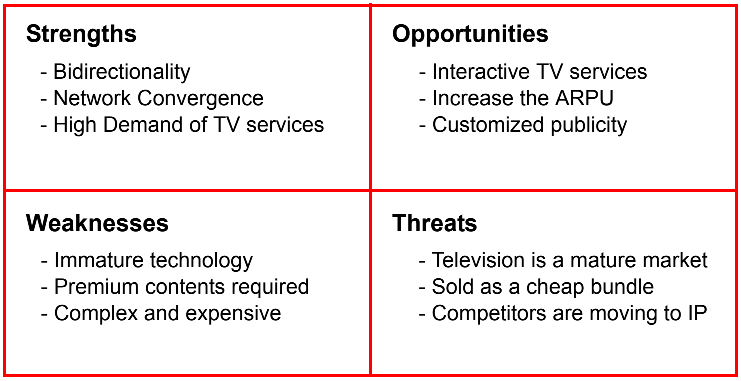
the delivery of television with the appropriate quality of service. *Second*, there is not bandwidth enough for standard or high definition TV. *And third*, there are a some protocols and multicasting methodologies that are not supported.

The underlaying network for IPTV distribution must be a converged network, IP centric, QoS enabled, and based on the multicast distribution and delivery of the television signals. This requirements today can only be fulfilled by privately managed IP networks.

3.2 Questioning the IPTV business models

Commercial video services delivered over a converged network should match on quality, price and contents, to the existing ones, because it is true that it exits a demand for TV, but this is a very mature and competitive market. Telcos must understand that this is a completely different case than when mobile telephony was launched, it was a totally new product and customers were willing to pay much higher fees. Mobile telephony was an extraordinary success story but, it is also true that 3G was a complete disaster, at least under the commercial point of view.

In order to minimize the possibility of a failure, telcos must define a business model that will help to maximize the chances of writing a successful story, by studying the strengths and weaknesses, plus opportunities and threats.



Strengths - Bidirectionality - Network Convergence - High Demand of TV services	**Opportunities** - Interactive TV services - Increase the ARPU - Customized publicity
Weaknesses - Immature technology - Premium contents required - Complex and expensive	**Threats** - Television is a mature market - Sold as a cheap bundle - Competitors are moving to IP

Figure 3.2 IPTV business model. Opportunities and Threats.

3.2.1 Strengths

IPTV provides a set of interesting possibilities thanks to the combination of digital television and internet-like features such interaction between the subscribers, network and services providers. This iteration is, at the end of the day, the key differentiator with the broadcast television.

3.2.1.1 Interaction

The audiovisual services based on IP have the maximum flexibility to combine television with interactivity. IPTV can be a simple replication of broadcast TV or Video, but can also add the interaction typical of internet. Interactive TV opens an interesting, and still unexplored, dimension that lies in between traditional TV and Internet.

Imagine how different the television can be if there is interaction between groups of subscribers, or between subscribers and content providers during the emission of TV magazines, or pay-per-view programs. For instance creating closed group of users that could interact each other sending messages while watching a favorite sport match in different locations.

3.2.1.2 Subscriber profiles

Operators may build user profiles that can define and change dynamically depending on user preferences. Experiences have demonstrate that there are different profiles of subscribers, some are passive, other look for specific contents, other enjoy interation. Then, it is possible to establish a direct relationship with the subscribers that would be totally impossible to do in broadcast TV (see Figure 3.1).

Customized information adapted for each profile will allow, for instance, each subscriber to receive reports about specific sport events, films or programs. It would be also possible to send publicity only to those customers matching specific profiles, this point is particularly interesting for advertisers since the impact of the publicity on TV has been declining.

3.2.2 Opportunities

Bidirectionality probably is the main strength of IPTV that can be transformed into the opportunity to develop new interactive applications involving all the active agents of the system. Interaction can create additional interest on users and could eventually be the way to capture more audience and increase profits, but this is not the unique opportunity for IPTV.

3.2.2.1 Customized Adverts

The interactive IP network enables a large number of opportunities. Moreover, the IPTV industry believe that targeted advertising based on subscriber profiles and interaction will generate the major source of revenues (see Table 3.1).

In IPTV the whole system is able to receive feedback about particular users preferences which is fundamental to customize adverts that will allow providers to increase publicity fees. It is possible to know exactly the preferences on program, audience peaks and duration. This will help advertisers to target better each segment of consumers. Furthermore, it can be possible to establish direct channels of communications between consumers, advertisers and service providers.

3.2.2.2 Unlimited Channels / Unlimited videos

Another interesting opportunity that the IP network can offer is the number of TV channels and copies of the same video that can be offered to the customers are theoretically unlimited. This fact is based on how the channels are distributed and videos are recorded an delivered. We will see how the IPTV distribution of programs is based on multicast agents, and the videos distribution is based on unicast addressing, depending on the type of application, but never a broadcast addressing mode is used. This strategy permits to exploit the resources in a very controlled way, maximizing the delivery of the information only to those customers that have subscribed it.

3.2.2.3 Personal services

Video and television market is changing, customers are willing to use personal video services (PVS) to save video clips in a network server or to transfer videos between PCs, TV devices, or mobiles, and IPTV is perfectly prepared to offer it.

Table 3.1
What do you think will be the principal revenue sources for IPTV?
(Source: Accenture)

	Minor	*Moderate*	*Major*
Network access fees	45%	40%	15%
Subscription fees for basic content	33%	40%	27%
Subscription fees for premium content	16%	44%	40%
Pay per view	19%	53%	28%
Advertising-mass market	30%	40%	30%
Advertising-targeted	16%	37%	47%

3.2.3 Weaknesses

To create real demand for audiovisual services delivered over IP will not be enough bundling television with broadband access and phone services. The long term success, requires to mature a business combining attractive contents with the interaction inherent to IP protocol.

3.2.3.1 Quality Contents

The access to quality contents is fundamental to enter and to stay in such a mature market that IPTV can facilitate. Telcos must acquire, not only new technical and business skills, but must also be involved in contents creation that is adapted to their specific consumer market. This explains why some telcos are creating their own studios or have signed contracts to find attractive programs, movies and specialized productions from film and program makers studios.

3.2.3.2 Complex and expensive

Interaction increases the complexity of the middleware, and the user interfaces as well, consequently software and the whole platform may become difficult to be used and unstable which would be even worse. In other words, more investments on middleware does not necessarily means more revenues (see Figure 3.3).

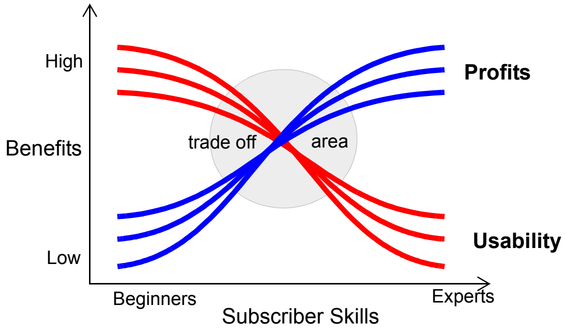


Figure 3.3 IPTV bidirectionality opens new opportunities to create new applications, whenever the user interface continues being simple and the middleware stable.

3.2.3.3 High Definition TV

High Definition TV (HDTV) must be planned and be considered a natural evolution of the whole business rather than a differentiator since cable TV, satellite and terrestrial digital TV are already moving forward this direction.

3.2.4 Threats

The whole IPTV business is not mature. The main threat may occur if providers do not discover their specific market segment. If eligible subscribers buy IPTV only because it as a cheap bundle of multiple services that also include television then the whole triple play service will become quickly a commodity with tiny margins. A similar case to ADSL, where self installing kits are common to save installations costs.

3.2.4.1 Profitability

Assuming that operators can get an important mass of subscribers and supply them with the proper contents and interactive facilities, is not clear yet that it can always become a profitable business. Research on the economics of mature cable markets indicate that about a third of cable networks deployed during the last decade are not profitable. How can we assure that IPTV will be?

If IPTV is successful traditional suppliers will react offering lower prices, consolidating companies and improving the services in quality and contents.

3.2.4.2 Intellectual Property Rights

The bidirectionality and digital downloading application can also damage the Intellectual Property Rights (IPR). The impact of piracy and the person-to-person video transfer can jeopardize the whole business. Content providers, operators and regulator must define the strategy to overcome this issue controlling customers use of IPTV facilities and legal regulations.

Table 3.2
"What are the business benefits your company hopes to achieve through IPTV business?" "Select up to two benefits" (Source: Accenture)

	Value	*Moderate*	*Major*
Reduce churn	13%	**Increase broadband**	21%
Acquire new customers	28%	**Adoption new services**	17%
New revenues	43%	**Develop new competencies**	15%
Drive profit growth	19%	**Respond to competition**	11%

3.3 REGULATORY FRAMEWORK

We have explained how IPTV is made up as a combination of television and telecommunications, two areas that in the past have been regulated independently. TV has always been a one-way communications. It was question of radioelectric spectrum, IPR protection, and control of several aspects regarding commerce, ethic, politic independence, education, language, and religious orientation. On the other hand, the Internet still remains very unregulated except on contents when it can be considered as criminal offences and certain aspects that apply for the electronic commerce. Internet regulations focus, in principal, on how to guaranty competition, quality and availability.

Table 3.3
Audiovisual service models

	Resolution	*Access*	*Device*	*Interaction*	*Benefits*	*Issues*
Cinema	Maximum	Walking	Camera + Wall	No	Last titles, Social event	Cost, time
Video Shop	Standard/High	Walking	DVD + TV	Start/Stp/Pause/ Fwd/ Bwd, Idiom	Large Archive, Privacy	Cost, time
Commercial TV	Standard	Wireless	Antenna + TV	No (idiom perhaps)	A lot broadcasters, Free	Ads, poor contents
Satellite TV	Standard	Satellite	Dish+ STB+ TV	No (idiom perhaps)	Wide area, Premium contents,	Installation
Digital TV	Standard/High	Satellite, Cable, TDT	Antenna/Dish/STB+TV	No (idiom perhaps)	Contents	Cost
IPTV	Standard/High	Copper, Fibre, FTTx	STB+TV/PC	Internet-like, group of users	Highest flexibility, All possible	Installation, cost
Personal Video	Low/standard	Copper, Fibre, FTTx	PC+Camera	Up/download	Sharing video, or archive	Server space

Table 3.3
Audiovisual service models

	Resolution	*Access*	*Device*	*Interaction*	*Benefits*	*Issues*
VoD	Standard/High	Copper, Fibre, FTTx	Server+STB+TV /PC	Start/Stp/Pause/ Fwd/ Bwd, Idiom	Privacy, anytime/anywhere	Installation, cost, IPR
Mobile TV	Low	Licensed Wireless	Mobile terminal	High	Anytime, anywhere, Mobility	Not ready
Internet TV	Low	Internet	Internet+PC	Download / streaming	No specific network	IPR control, QoS
Internet P2P	Standard	Internet	Internet+PC	Start/Stp/Pause/ Fwd/ Bwd, Idiom	Download to watch, low cost	IPR control, Bandwidth
Personal Video	Low/standard	Copper, Fibre, FTTx	PC+Camera	Up/download	Sharing video, or archive	Server space

So far IPTV is being deployed in a similar way as the Internet, but competitors would prefer a similar regulations in line with the TV broadcast. Obviously the IPTV industry is afraid that regulations could compromise the growth of this new technology.

Broadcasters, whatever they are, feel that IPTV has the potential to be an important threat to their business. The worse of the nightmares of broadcasters is to see subscribers controlling their own TV and video channels, choosing what programmes and events they want to watch and download their favorite contents, in a very simple, free and easy way. The question is: Can broadcaster suffer similar consequences as the music industry suffered after Internet success?

Independly of competition controversies, the IPTV industry is entering in a regulated content. The future regulation may eventually distinguish between linear and non-linear services:

- **Linear**, such as internet-TV, the provider decides when the program is transmitted, streamed or web-cast.
- **Non-linear**, such as video on demand, subscribers decide when and where the program is downloaded.

Linear services are generally subjected to regulations that also apply to broadcasters regarding IPRs, and the sort of contents, while non-linear services have similar IPR regulations but may enjoy less control regarding religion, politics, values, violence, or adult contents. Nevertheless non-linear services have increasingly more legal limitations as, in some markets, many content providers do not allow the storage of their programs and there exists regulations to control what a personal video recorder can do when connected to, or installed in, an IPTV network.

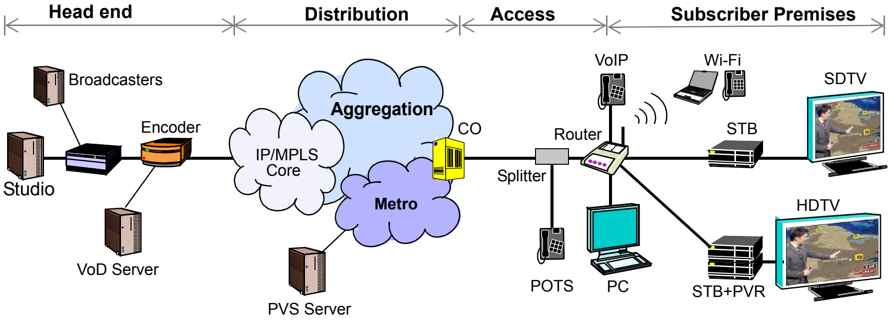


Figure 3.4 Architecture to support audiovisual services including IPTV, VoD and PVS. It is remarkable the IP bidirectionality that allow subscribers interaction with the television network.

3.4 ARCHITECTONIC DESIGN

The triple play bundle, and particularly IPTV, is a recent achievement thank to the evolution of IP architectures that can support differentiated QoS. The traditional bandwidth bottleneck at the first mile does not exist anymore, as there are a growing number of alternatives capable to deliver real-time TV. Finally it is also remarkable the evolution of codecs like MPEG that have made possible more compression, and resolution, squeezing much better the available bandwidth.

In many aspects IPTV roll out is probably the most important challenge telcos have assumed on this decade, because the project requires to deploy a complete new distribution network, renovate the access together with the installation of new servers, and customer devices.

3.4.1 Television services roll out

The challenge of rolling out compelling audiovisual services is not an overnight project but a process to be achieved in consecutive steps:

1. **Define the business model**. The analysis of the market and the competitors to determine key parameters as service charges and size of the market. It will also be convenient to think on how to define service differentiators.
2. **Architectonic design.** Defining which network elements and resources are required at each network including the contribution side, aggregation and access network.
3. **Service definition**. It must be described in terms of picture size, coder and

decoder standard, frame formats, encapsulations, and video profiles.

4. **Equipment approval**. Once the business model is clear it is necessary to select the type of servers, network nodes, and customer devices that best matches the technical requirements and the available budget.
5. **Network commissioning**. The infrastructure roll-out require the verification of IP continuity and network performance, that will permit engineers the establishment of routes, and to define multicast nodes that will control the routing and the quality management.
6. **Service Provision**. Video servers and subscribers equipment must be connected, and applications installed to check end-to-end performance and the capability of the network to deliver differenciated video services under real traffic conditions.
7. **Service assurance.** The promised service may be affected by a number of issues that would modify the initial quality conditions. In is necessary to identify and troubleshoot the causes of a bad video quality affecting customers and expressed as subscribers quality of experience.

3.4.2 Business model definition

A lot of the major operators across the world are deploying converged network to support video distribution simultaneously with other services like data transport, internet or telephony. The subscriber access network should scale from current bit rates up to a level enough to support at least one TV channel to target different segments of the TV market like film enthusiast, sport aficionados, children channels, live shows, adult contents, or holiday makers.

But IPTV requires a new and complex architecture that does not have a unique solution. To achieve good results under the technical and financial point of view it is necessary to plan and deploy a new architecture considering a number parameters that depend on factors like market conditions, and resources availability, in order to define the business model:

- **Number of programs.** Each provider have to define the selection of programs, that will make available to the subscribers. All of them must be able to reach every subscriber (see Figure 3.5).
- **Image characterization.** Define the quality of the video in terms of number of pixels or frames per second. Generally speaking the better the quality of the image is the more bandwidth is required (see Table 3.5). There is a move to high definition TV (HDTV) it is interesting to realize that manufactures are already selling a lot of receivers despite of the lack of HD programs.
- **Coder and Decoders**. Several are the alternatives to compress and code programs including the MPEG family of standards, and also the Microsoft alternative. The differences are important, elder standards like MPEG-1 are the less

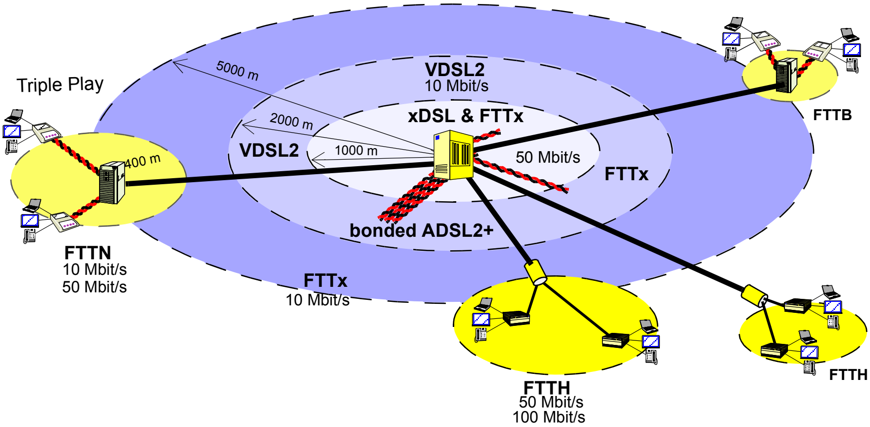


Figure 3.5 Several access alternatives allow the bit rate escalation from a few Mbit/s up to 100 Mbit/s for multiroomed HDTV.

efficient. Others such MPEG2, have an good installed base but progressively substituted by the latest standards like MPEG4 and WM-9 which are more flexible and efficient (see Table 3.5).

- **Unicast and Multicast traffic**. IP television generates multicast and prioritized traffic, while video on demand services and Internet access generate unicast and best-effort traffic. It means that the network must support a combination traffics, and depending on what the proportion is, the architecture may be different. Generally the higher proportion of unicast services the more resources and bandwidth the network requires.
- **Number of subscribers**. IPTV requires costly infrastructures then density of users is a criteria to plan new deployments that are always cheaper in metropolitan areas than in suburban or rural areas. It makes sense to focus first on potential segments that may be less reluctant to contract new services like those that already have broadband access.
- **Traffic patterns.** The knowledge of traffic patterns may be used to modify dynamically the topology of the network reassigning resources to adapt the network to traffic conditions. Continuous monitoring will help to discover picks and valleys of traffic repeated daily and weekly. Even seasonal patterns can be shaped. Unicast traffic, typical of video on demand and personal applications, is more distributed across the time but it also follows more unpredictable patterns than the multicast traffic characteristic of IPTV.
- **Investments and budget**. Regardless of the technical considerations it is always of key importance the financial capacity, the value on market of the busi-

ness, and the terms to recover the investments. IPTV should not go to price battles against TV broadcasters, as bidirectional TV requires a considerable number of new infrastructures that it is reflected on higher CAPEX and OPEX.

The detailed analysis of these parameters will help in the process of taking decisions that define many aspects of the business case including architectonic like the head-end, distribution network, access network and subscriber premises equipment (see Figure 3.4).

3.4.3 Head-end

The head-end or contribution network is the place where the programs are coded, compressed, multiplexed and finally streamed in IP packets using the most convenient physical interfaces. Video and audio content is typically compressed using either a MPEG codec to define the presentation parameters, such as pixels per line and frame refreshing, to playback the video signal. and then streamed through IP networks in multicast or unicast addressing modes.

This scheme may no be exclusive of telcos and new entrants, since there is a trend of using IP packets everywhere, including in satellite and cable networks. Broadcasters are willing to upgrade their networks to two-way capability in order to embrace all the advantages of combining internet concepts with digital television.

3.4.4 Distribution network

The distribution network has to dispatch the audiovisual signals across regional and metropolitan networks until reaching the customers premises. A high capacity core, assisted with protocols like RTP or RTSP, must guarantee every IP packet to be multicast or unicast without degrading the quality parameters such as delay or packet lost. IP Multicast is a method in which information can be sent to multiple, but not all, nodes or hosts at the same time (see Figure 3.6).

Distribution networks are IP centric, and generally based on Carrier Ethernet which is responsible of the traffic aggregation. Architectures supporting isochronic applications require excellence in design and management which is often achieved with engineering tools such as MPLS or VPLS that can transform a native best effort protocols (which is the case of Ethernet and IP) into a quality assured network. Different polices of prioritizing like Diffserv can be implement to keep in good shape the quality of the TV signal.

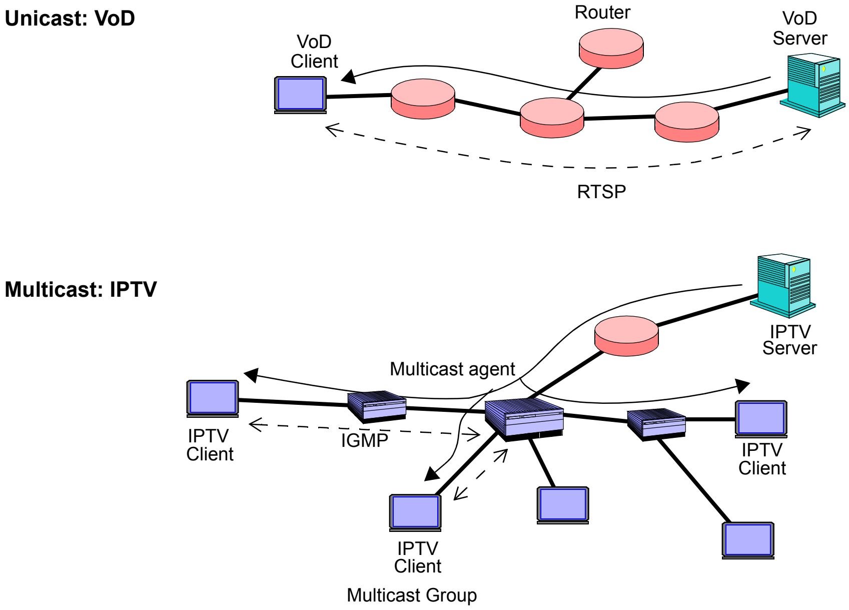


Figure 3.6 Video on demand is delivered as a unicast service and managed by the RTSP protocol, while IP television is a multicast services managed by the IGMP protocol. This approach has proven to be and efficient and very scalable solution to deliver signals simultaneously to a large number of subscribers.

3.4.4.1 **Access network**

The access network spans the last mile providing a link between the service provider and the subscriber site. Several technologies available including ADSL2, ADSL2+, VDSL2, FTTN, FTTH, and WiMAX. The selection will depend on parameters like the necessity of bandwidth, the distance and the financial restrictions (see Figure 3.7).

However, despite of the great progress on access technologies occurred during the last decade, subscribers, due to the bandwidth limitations, are able to receive only two or three channels simultaneously if they have a good access, otherwise they would receive only one channel. The zapping function is a request to the network, using the Internet Group Membership Protocol (IGMP), to notify the program selected to be received, which in technical terms means that the subscribers wants wants to be part of the multicast group that contains the selected program.

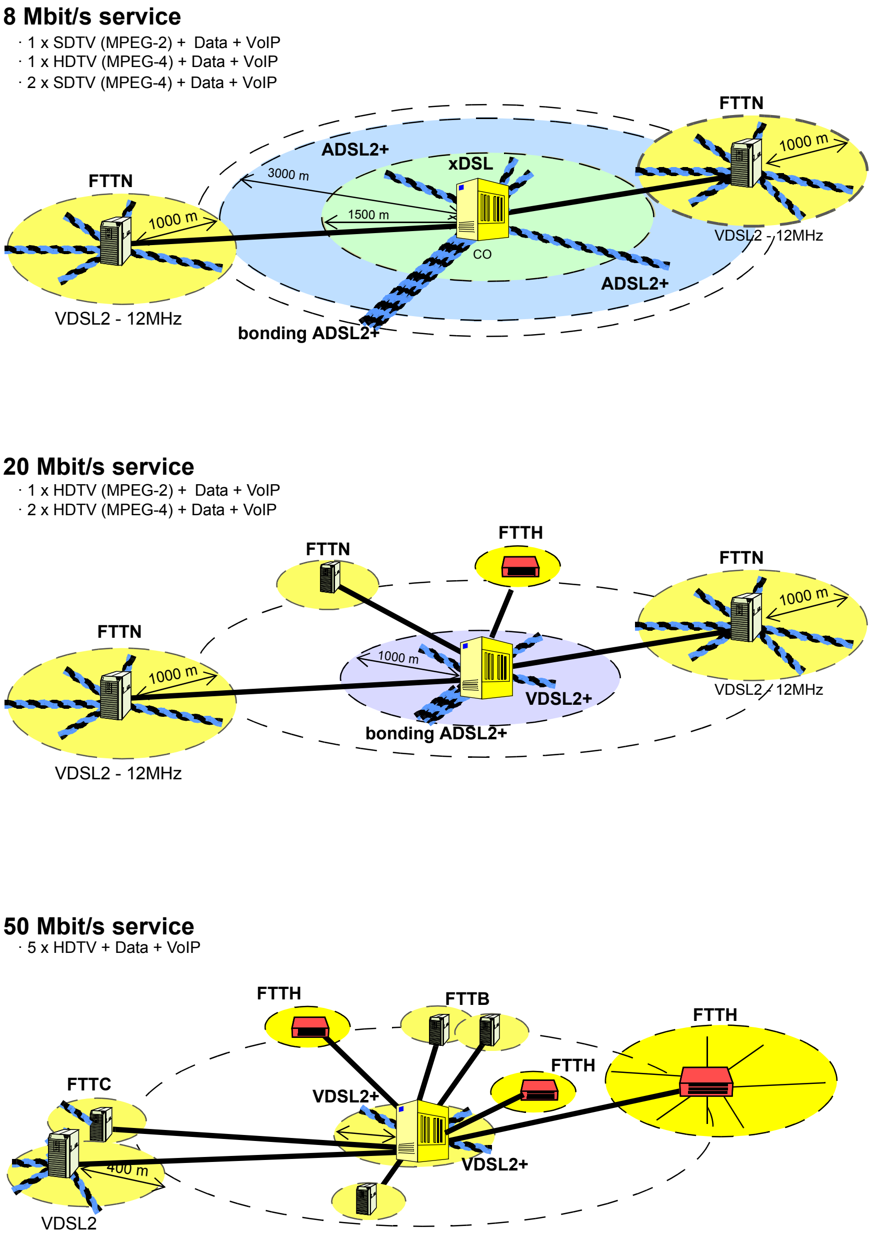


Figure 3.7 The access technology depends on the service to be rolled out, the distance to the central office, and the available budget. Copper loop is everywhere available but has bandwidth limitations compared with fiber optics which is future proof.

3.4.5 Subscriber Site

The IP packets finally reach the receiver site as a video stream. The IPTV play-back requires either a personal computer or a Set Top Box (STB) connected to a TV, other gadgets like PDA and Mobile hand-sets could also be possible. The interaction between the subscriber and the network is done by means of protocols like the Internet Group Membership Protocol (IGMP) for channel selection in the case of IPTV and the RTSP to manage the VoD applications.

The STB is the central and most relevant element at the customer site, the point where many features must be supported. The STB should be flexible enough to be upgraded as soon as new facilities or standards become available to avoid its replacement which is always an expensive operation. For instance some STB support more that one compression algorithms, others have hard disk for recording programs, some allow to watch a program while recording another, and the best ones can even record several channels simultaneously. In a few words, STB features are so significant for the business development that must be taken into account onto all strategic road maps to develop the IPTV business.

3.5 Television and Video Services and Applications

The new IPTV services can be seen as the combination of television with the highly interactive of Internet concepts. The result is a number of new applications based on the delivery of audiovisual contents in a bidirectional, customized and controlled way.

3.5.1 IPTV protocols

The support of multiple data and multimedia applications over a common infrastructures probably is the main achievement of converged networks. It is only above the layer three or four where we can find protocols devoted just for only specific applications (see Figure 3.8) below are the same. This is the case of IGMP a protocol intended to support IPTV channel zapping, or the RTSP that provides VCR features (i.e. start/stop/fwd/bwd) specifically devoted to video on demand applications.

Regarding the codecs used for video digitalization, compression and transport, several options are available such as MPEG-4 or VC-1. It is difficult to say which the best option is, it all depends on the business model, strategic decisions or simply the availability of standards at the moment when the service is rolled-out.

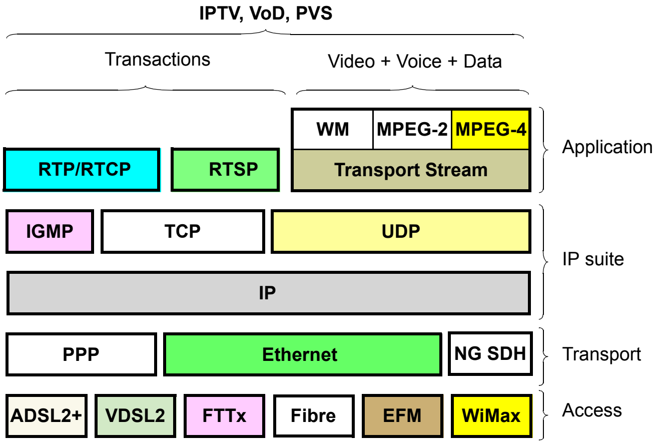


Figure 3.8 Protocols used for audiovisual services over converged networks.

3.5.2 Video on Demand Services

The Video on Demand (VoD) applications are a bit different to live IPTV since it requires the establishment of a point-to-point unicast relation between the server and the subscriber to enable subscribers to select and watch an stored video interactively across the IP network. VoD systems have two important features:

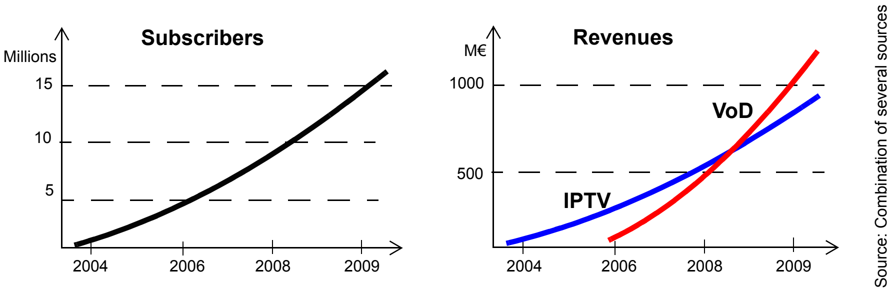
1. Subscribers can choose the video they want to watch from a digital library, deciding the moment and the place they want to start.
2. Typical VCR/DVD functionality are provided by the RTSP protocol including *start, pause, stop, forward, backwards, fast forward,* or *fast rewind* (see Figure 3.30).

There are many different implementations of VoD, that can be classified in two groups, depending if the contents is streamed or it is downloaded:

- **Streaming VoD,** real-time service the video is downloaded directly from the server to the TV, consequently QoS transport requirements are similar to IPTV. A lot of bandwidth and processing capacity are necessary at the head-end and across the distribution network.
- **Downloading VoD,** best-effort service, the video is downloaded and saved before being decoded and displayed in a similar way to those popular person-to-person applications on the Internet. The hard disk where video is saved can be at the provider's network or at customers site.

Whatever the implementation is, VoD can be the most relevant source of revenues of the Triple Play bundle. The number of stored programs such as films, video-clips, sport events, etc. can be so massive that traditional video shops can't match and the top ten most popular titles will never become unavailable as they are just downloaded.

Definitely video-on-demand service is one of the killer applications, since Cable and Satellite broadcasters have more difficulties in implementing this highly interactive service



Source: Combination of several sources

Figure 3.9 IPTV business, some market forecasts for Europe.

3.5.3 Personal Video Recording Services

Personal Video Recording (PVR) services dispenses the users more flexibility to watch TV. Each subscriber has his own storage disk at the network, or at the subscriber site, to record and remove programs that will be playback at any moment according user convenience just using VCR-like functions.

The are always programs and events that subscribers would like to record to replay in the future. Also consider the case when a customer misses a multicast program, total or partially, but he/she would be willing to record it. Think how normal this circumstance is, when the subscriber is watching a live multicast program and then he/she wants to *pause* for a while, or to *skip* the adverts section, an then *go back* to some point during live transmission.

PVR service includes Networked Personal Video Recorder (NPVR), Client Personal Video Recorder (CPVR), Time shift TV, and Last week TV.

Networked Personal Video Record

NPVR is a consumer service where real-time IP television is captured in a network. When using NPVR subscribers can record programmes in the network-based server, when they want, without needing yet another device or remote control. The recording is performed on the network server so that subscriber's devices are totally independent of the storage process.

It is important to take into account the technological limitations which can result in noticeable delays during playback, and in response to the "pause", "rewind" and "fast forward" features that it shares in common with VCRs and DVD Players.

Client Personal Video Record

Many people would still prefer to have their own PVR device, as it would allow them to choose exactly what they want to record, and eventually to create their own library of DVD.

Private TV

Groups, enterprises, and associations on specific topics can create their own TV uploading their videos and programs to be downloaded by VoD or even scheduling the transmission to a closed group of subscribers. This application transfer the control of the contents to the subscribers while the TV operator only provides the architectures and the management of whole system.

Last week TV

The IPTV operation may offer the complete retrieval of last week (or any other period of time) programs. This service made available to subscribers that missed the broadcasting schedule.

Time Shift TV

Time shift TV is a feature that combines video recording of real time TV streaming. The subscriber watches the program with certain delay allowing to pause for a while, or to skip the advertisement section, go back to some time point during live broadcasting. There is no intention of recording the whole program just of doing a kind of buffering of live emissions to allow this flexibility.

3.5.4 Converged Telephony

IPTV can be easily converted into an integrated telephonic system supporting video conferencing that can be a real differentiator. Video conferences have not been very popular despite of the effort made by legacy technologies like ISDN, ATM and 3G. Now video conferences have a new opportunity that can be successful finally because it will be low cost and target users will be part of a wider bundle that include many different services.

Supplementary services such as the call id can be integrated on the TV screen and messaging can also be part of the application that includes TVs, PCs and mobile phones. We should bear in mid that Triple Play is often seen as a defensive strategy to keep mobile and fixed line customers, upgrading the service and integrating it in new platform looks on that direction.

3.6 Formats and Protocols

3.6.1 Analog TV

Broadcast analog TV was implemented differently across the world. In Europe 625 lines are displayed at a frame rate of 25Hz, while at the US and Japan the systems are made of 525 lines at a frame rate of 30Hz.

Initially the TVs only had luminance, represented by Y, to describe the brightnesses transition from back to white. When color TV was developed it was necessary to add chrominance by means of two more components, U and V, resulting in the YUV model. The RGB (Red, Green, Blue) signals coming from a color video camera, can be transformed directly:

$$\textit{Luminance } (Y) = 0.299R + 0.587G + 0.114B,$$

$$\textit{Chrominance } (U) = 0.493 \times (B - Y)$$

$$\textit{Chrominance } (V) = 0.877 \times (R - Y)$$

The YUV model was adopted quickly since it permitted the compatibility between the old black and white, and the new color television systems. Note that to represent a color image in a b&w device is only necessary to use the luminance while chrominance components are discarded making U = V = 0.

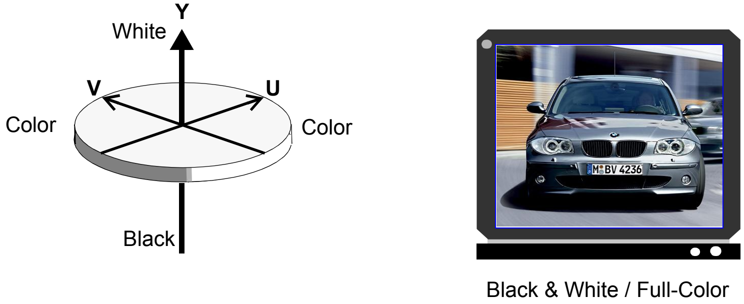


Figure 3.10 YUV model was invented to separate color and luminance to permit the compatibility of color, and black-and-white TVs. YUV coding converts the RGB signal to an intensity component (Y) that ranges from black to white plus two other components (U and V) which code the color. The conversion from RGB to YUV is linear, without loss of information.

3.6.2 Digital TV

The next milestone in the TV evolution was the digitalization that makes easier the control, storage and distribution of the audiovisual streams, while simplifies the quality management until the signal is delivered.

The ITU-R recommendation BT.601, described how to digital the analog YUV signal into 720 luminance samples and 360 chrominance samples per line. Samples become into pixels to create the picture which is refreshed 25 times per second (see Table 3.4). This recommendation indicates how to sample and quantizes the signal to obtain a binary sequence of pixels. The digitalization maintains the three digits of the YUV model now transformed:

- one number (Y') for the Luminance.
- two numbers (C_B, C_R) for the chrominance.

3.6.2.1 Chrominance subsampling

Human vision is less sensitive to color than bightness, then chrominance is able to be subsampled at lower rate than luminance without severely destroying the image quality. Subsampling results in a significant bandwidth reduction.

The sampling scheme can be expressed with a notation of three digits A:B:C where A indicates the luma sampling reference, in our case A is always equals to four. B represents the chroma horizontal sampling factor relative to A. And finally C represents the chroma vertical sampling factor relative also to B (see Figure 3.11). Here we have three cases:

- **4:4:4** means that all luma and chroma components are vertically and horizontally sampled at the same rate. If luminance is digitalized at 720x576 samples, the chrominance have also 720x576 samples (see Table 3.4).
- **4:2:2** means that the two chroma components are vertically sampled at the same rate of luma rate, while horizontal sampling is done at the half rate. For instance if luminance is digitalized at 720x576 samples, the chrominance are at 360x576 samples (see Table 3.4).
- **4:2:0** means that chroma is horizontal and vertically sampled at the half rate of luma sampling rate. For instance if luminance is digitalized at 720x576 samples, the chrominances are at 360x288 samples (see Table 3.4).

Table 3.4
TV digitalization in a 720x576 pix definition, using 4:4:4, 4:2:2 and 4:2:0 models

		pix/line (horizontal)	*lines/frame (vertical)*	*frames/s*	*bits/pix*	*Mbit/s*	***Total Mbit/s***
4:4:4	**Luma Y'**	720	576	25	8	82.944	
	Chroma C_B	720	576	25	8	82.944	**248.832**
	Chroma C_R	720	576	25	8	82.944	
4:2:2	**Luma Y'**	720	576	25	8	82.944	
	Chroma C_B	360	576	25	8	41.472	**165.888**
	Chroma C_R	360	576	25	8	41.472	
4:2:0	**Luma Y'**	720	576	25	8	82.944	
	Chroma C_B	360	288	25	8	20.736	**124.416**
	Chroma C_R	360	288	25	8	20.736	

3.6.3 Audio and Video Codecs

There are a number of specifications of codecs (coder-decoder) for of audiovisual signals that define the coding, compression, and streaming the audiovisual contents. These standards facilitate the independent implementation of interoperable encoders and decoders.

The most popular family is MPEG (Moving Pictures Expert Group) defined by the ISO/IEC and ITU-T. Windows Media, which basically is a Microsoft development, is also an interesting codec that is being considered by many of the new IPTV entrants.

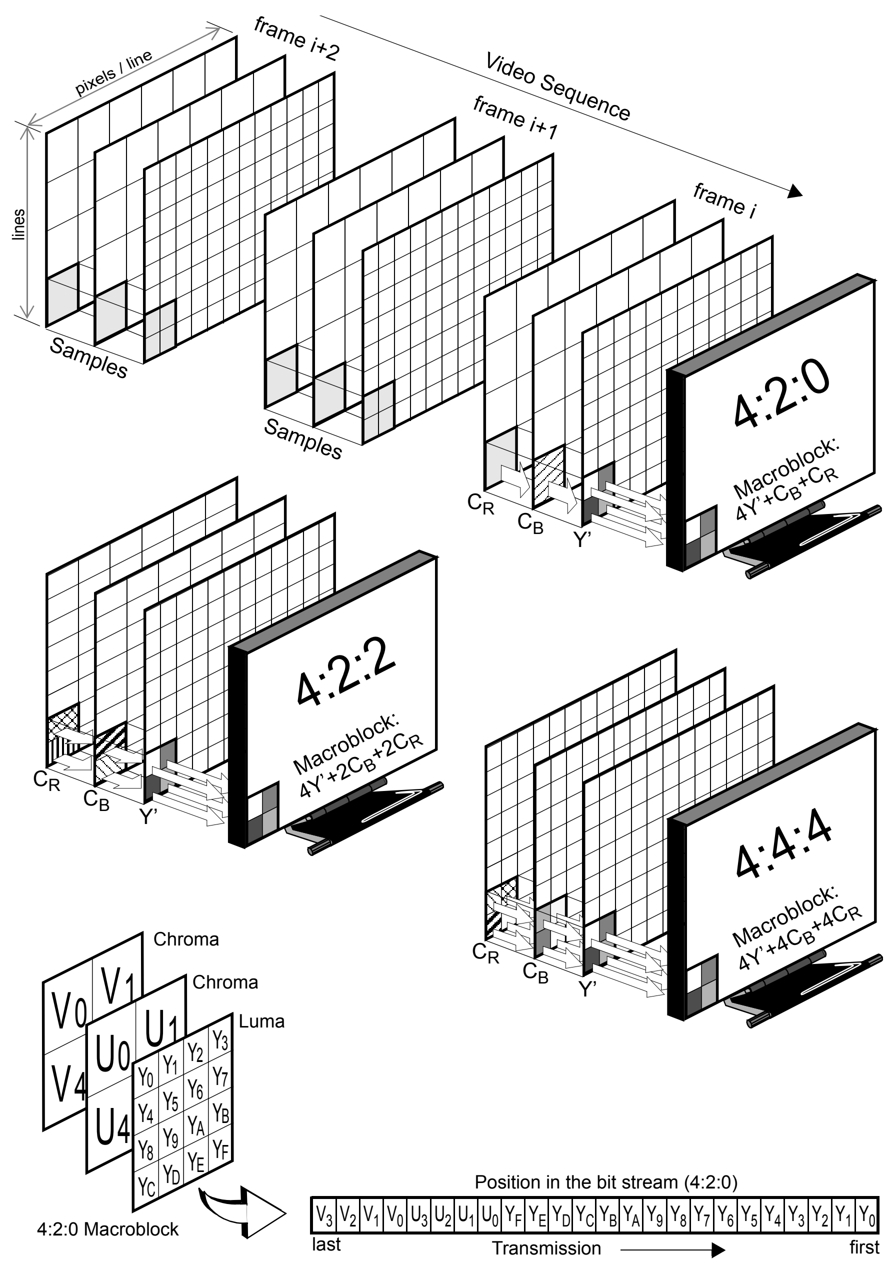


Figure 3.11 The YUV video signals can be digitalized according several schemes to reduce the bandwidth. The 4:4:4 scheme has the same samples for luma, and chromas, the 4:2:2 subsamples chromas horizontally, while the 4:2:0 subsamples chromas horizontal and vertically.

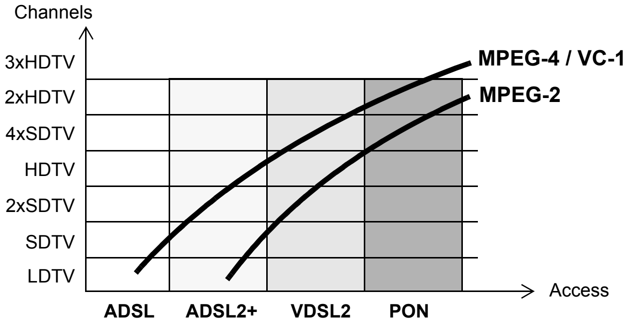


Figure 3.12 The access technology and the codecs determine the quality and the number of channels of the IPTV service.

- **MPEG-1**, published in 1993 was the first digital video standard codec that made possible the evolution from the analog to the digital dimension regardless of the analog standards. MPEG-1 compress and encodes audio and moving at rates around 1.5 Mbit/s, providing a video resolution equivalent to VHS cassettes, targeting the first generation of audio and video CD-ROM applications. MP3, the audio part of MPEG-1, has become very popular thanks to the Internet, the digital radio and the new and supports for digital audio.
- **MPEG-2 (ITU-T H.262)**, published in 1995 is a superset of MPEG-1 providing a wider range of bit rates from 2 Mbit/s to 20 Mbit/s, several levels of quality, and video screen resolutions. MPEG-2 application have been very popular since mid 90's and is being used in Satellites, Cable, DBS, DVD, and early implementations of IPTV.
- **MPEG-4 part 10 (ITU-T AVC/H.264)**, published in 1999, is very flexible codec which has a range of bit rates from 5 kb/s to 10 Mb/s making it suitable for mobile video, standard definition and high definition TV. MPEG-4 can save up to the 50% of the bandwidth then new entrants on IPTV have selected it while existing MPEG-2 applications are planing the migration to MPEG-4.
- **SMPTE VC-1 (WM9V)**, is a video codec specification that has been standardized by the Society of Motion Picture and Television Engineers (SMPTE) and implemented by Windows Media 9 (WM-9) which has similar features than MPEG-4, but probably will permit an smoother integration with PCs or hybrid PC-TV devices.

Table 3.5
Bandwidth of Standard and High Definition TV

	Screen size	*MPEG-2*	*MPEG-4 Part 10 (H.264)*	*VC-1 (WM9)*
SDTV	704 x 480	3.5 Mbit/s	2-3.2 Mbit/s	2-3.2 Mbit/s
HDTV	1920 x 1080	15 Mbit/s	7.5-13 Mbit/s	7.5-13 Mbit/s

3.7 How a codec works

The Moving Picture Experts Group (MPEG) is a working body within the ISO that is responsible for developing video and audio encoding, compression and standards for digital television delivery and multimedia digital video applications.

A detailed explanation about codecs goes far beyond the objectives of this book. Nevertheless, in order to illustrate the video compression techniques we have chosen MPEG-2 as it has been a milestone for the other codecs such as MPEG-1, MPEG-4 and VC-1 that often use very similar techniques.

3.7.1 MPEG-2 Levels and Profiles

MPEG-2 is a intricate standard intended to support a wide range of applications. It is very flexible permitting implementations with different resolutions and qualities for making it suitable for DVD, Satellite, Cable and IP Television. However, in order to simplify and to reduce the cost, commercial devices do not need to implement the full MPEG-2 standard but the subset that matches the requirements of the TV system. Then the selected subset is defined by two descriptors, the profile and the level:

- The *level* describes the constrains on parameters that manage the video presentation and streaming (see Table 3.6). Four different levels have been defined.
- The *profile* defines a subset of algorithmic tools. Five profiles are available to achieve different layers of quality improvements (see Table 3.7).

The exact combination of profile and level, exactly describes the MPEG-2 subset of features to which encoders and decoders are required to implement.

Table 3.6
Combined constrains, levels, and applications.

		Constrains					
		Picture-Size		*Video Streaming*			
		Max. Lines/Frame	*Max. Pixels/Line*	*Buffer Size*	*Max. Frames/s*	*Max. Mbit/s*	**Applications**
Level	*High 1920*	1152	1920	9,781,248	60	80	HDTV
	High 1440	1152	1440	7,340,032	60	60	Entry HDTV
	Main	576	720	1,835,008	30	15	SDTV
	Low	288	352	475,136	30	4	Low entry TV

3.7.1.1 MPEG-2 Levels

Levels describe the constrains on video presentation and video streaming parameters. Constrains are upper limits on what codecs must operate below. The four levels defined are High 1920, High 1440, Main and Low each one designed to fulfill requirements of different applications in terms of picture size, bandwidth and buffer size limitations (see Table 3.6).

Table 3.7
MPEG-2, combination of profiles and levels.

		Profiles				
		Non-scalable		*Multilayer-Profiles*		
		Simple	*Main*	*SNR*	*Spatial*	*High*
Levels	*High 1920*		**4:2:0** 80 Mbit/s			**4:2:0 / 4:2:2** 100 / 80 / 25 Mbit/s
	High 1440		**4:2:0** 60 Mbit/s		**4:2:0** 60 / 40 Mbit/s	**4:2:0 / 4:2:2** 80 / 60 / 25 Mbit/s
	Main	**4:2:0** 15 Mbit/s	**4:2:0** 15 Mbit/s	**4:2:0** 15 / 10 Mbit/s		**4:2:0 / 4:2:2** 20 / 15 / 4 Mbit/s
	Low		**4:2:0** 4 Mbit/s	**4:2:0** 720x576 4 / 3 Mbit/s		

3.7.1.2 MPEG-2 Profiles

There are five profiles defined. The three more sophisticated are capable to deliver several qualities in a multilayer structure, while the other two are orientated to simpler implementations.

- *Simple*. No-scalable profile suitable for symmetric and time sensitive applications like videoconferencing that requires delays below 150 ms. To achieve that it does not use B-frames (see Paragraph 3.7.2.4) to avoid reordering and buffers, then delays are minimized.
- *Main*. No-scalable profile suitable for asymmetric applications like IPTV what makes that most decoders support main profile and main level. Main profile manages all types of frames: I, P and B, then it is more efficient but adds about 120 ms delay for frame reordering.
- *SNR*. Scalable profile that supports two layers of enhancements. The lower layer transmit the stream to a specific quality while the upper contains video refinements, allowing on this way two qualities simultaneously. This profile has been suggested to be used in Terrestrial Digital TV.

- *Spatial.* Scalable profile which has been suggested for HDTV emissions compatible with standard resolution receivers. Spatial profile is characterized by images coded and streamed at different resolutions. Lower layer carries low resolution images while higher improves with higher resolution.
- *High.* Scalable profile that supports three layers of consecutive enhancements to achieve on this way maximum flexibility and different qualities. The high profile supports the for the 4:2:2 scheme (see Paragraph 3.6.1) and includes both, the SNR and Spatial enhancement tools.

3.7.2 MPEG Compression

MPEG reduce the size of the video stream compressing them it squeeze resources such as bandwidth or disk, that are the media to transport and store video signals but without compromising the quality. In some respects, compression is a trade off between the bandwidth required and the quality obtained (see Figure 3.12). Compression technics are generally based on human perception and mathematical algorithms. In particular MPEG uses the following techniques (see Figure 3.13):

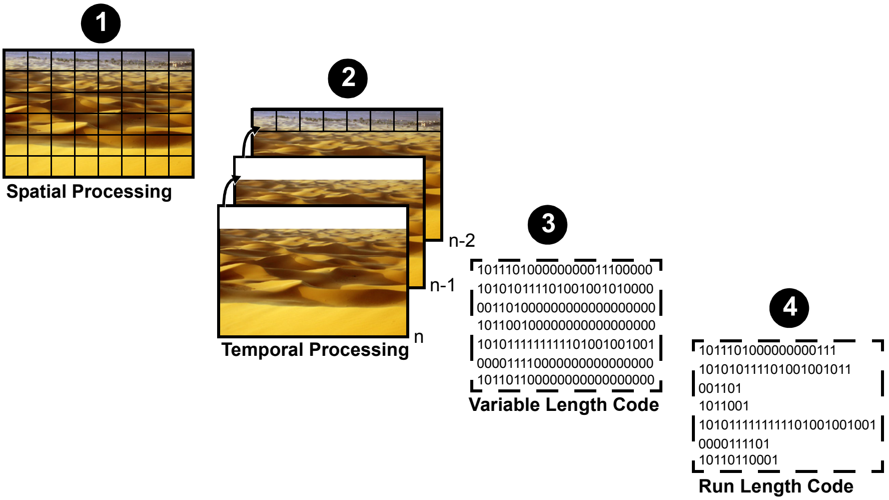
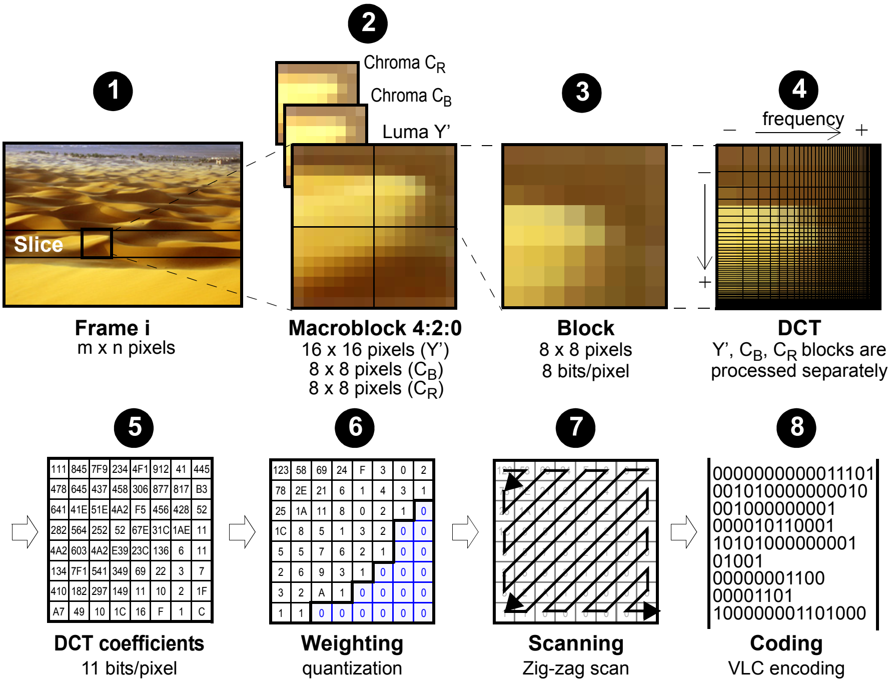


Figure 3.13 The four steps MPEG-2 compression four steps.

1. *Spatial Processing or Intraframe coding*, using Discrete Cosine Transform (DCT) to remove shapes the human eye can't see. This technique is based on the fact that pixels are not independent but correlated in color and brightness with the neighborhood.
2. *Temporal Processing or Interframe coding*, which looks for redundancies

between consecutive frames and does prediction of moving blocks. This compression takes advantage of the fact that may areas of one frame are the same to the previous one frame, the some areas are predictable i.e. backgrounds that tend to be repeated across several frames.

3. *Variable Length Code (VLC)*, which uses shorter encoding to reduce size, this a mathematical process to reduce the size of the digital code.
4. *Run Length Encoding*, replacing large sequences of zeroes to reduce the length of the digital stream, this a another mathematical process.



111	845	7F9	234	4F1	912	41	445
478	645	437	458	306	877	817	B3
641	41E	51E	4A2	F5	456	428	52
282	564	252	52	67E	31C	1AE	11
4A2	603	4A2	E39	23C	136	6	11
134	7F1	541	349	69	22	3	7
410	182	297	149	11	10	2	1F
A7	49	10	1C	16	F	1	C

123	58	69	24	F	3	0	2
78	2E	21	6	1	4	3	1
25	1A	11	8	0	2	1	0
1C	8	5	1	3	2	0	0
5	5	7	6	2	1	0	0
2	6	9	3	1	0	0	0
3	2	A	1	0	0	0	0
1	1	0	0	0	0	0	0

```
0000000000011101
001010000000010
001000000001
000010110001
10101000000001
01001
00000001100
00001101
100000001101000
```

Figure 3.14 Spatial processing is based on the fact that pixels that belong to a block are correlated then is it possible to remove the correlation. Luma and chroma are processed separately.

3.7.2.1 Spatial Processing

It is known that human eye is less sensitive for high frequencies of the picture shapes (do not confuse with the color-frequency concept) therefore, a DCT process is performed to eliminate or reduce the contribution of those components less significant on human perception. The process is as follows (see Figure 3.14):

1. The image is scanned horizontally in of 16 pixels wide slices. Slices shall not overlap, but the position of slices may change from picture to picture.
2. Each Slice is divided in macroblocks that contain block for the luminance and chrominance components. The number of blocks that vary depending on the luma/chroma scheme (see Figure 3.11). The 4:2:0 is made by six blocks (4Y'+C_B+ C_R), the 4:2:2 is made by eight blocks (4Y'+2C_B+ 2C_R), and the 4:2:0 is made by twelve blocks (4Y'+4C_B+4C_R).
3. Each block has 8x8 pixels, and each pixels is represented by 8bits.
4. A two dimensional DCT transforms each block into the frequency domain indicating the contribution of vertical and horizontal components. DCT concentrates the visual energy at the top left corner that corresponds to the lower frequency, our eyes most sensitive.
5. DCT transforms images into its frequential components increasing the coefficient for pixel size from 8 to 11 bits.
6. Weighting is performed according human perception: higher frequencies are the more coarsely quantized, therefore codification of higher frequencies is minimized while zero value is assigned to coefficients that may be below perception thresholds or provide a poor contribution to the image quality. Finally shorter codes are assigned for most common samples.
7. The non-zero values are then scanned to obtain the sequential binary stream, following a zig-zag scan to get first coefficients in the top-left corner, the most important for human perception, while the bottom right corner, the less important, are the last to be scanned.
8. The scanning results into a digital stream that tends to put zero coefficients together, then using Hoffman encoding a variable length code (VLC) that substitutes the non-zero coefficients by codewords of different length that are generated depending on the probability they can occur. This fact facilitates the final compression with zeros detection to be replaced by a code.

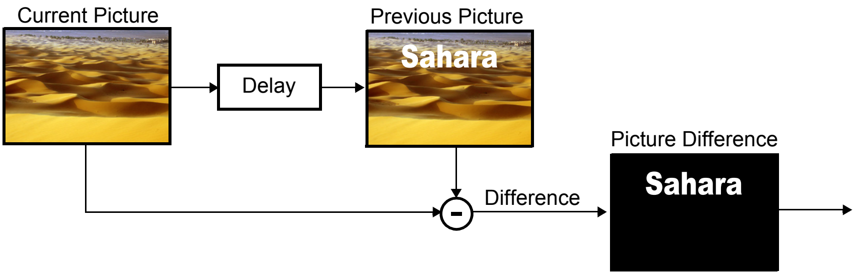


Figure 3.15 The difference between successive pictures is calculated by subtraction. The difference can also be spatially compressed using DCT. This method makes a good prediction for stationary areas such as backgrounds.

3.7.2.2 Temporal Processing

Temporal processing is based on the fact some areas of video are repeated across several frames during a period of time, i.e. stationary backgrounds. Then, instead of coding the same block again and again, the repeated block uses a reference to the place where it has already been represented.

The best interframe prediction are those referred to stationary areas that are repeated across several frames exactly in the same coordinates of the frames. Predictions can be either *backwards* if a previous block is used as reference, either *forward* if a late block is used as reference (see Figure 3.19).

A bit more difficult are the predictions of moving areas of the picture, that is, regions that have moved to another place of the picture sequence. For this cases a new algorithm known as motion compensated predictions is used.

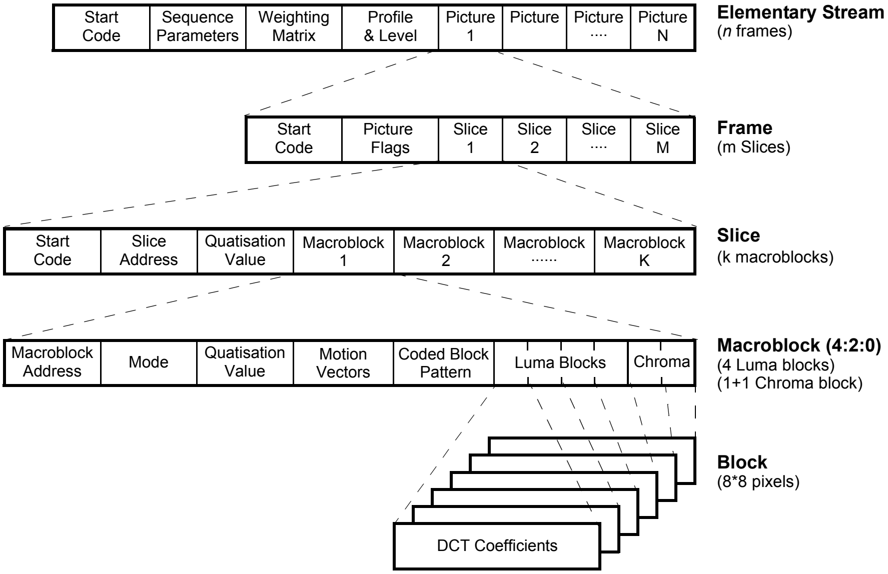


Figure 3.16 Elementary Video Stream (ES). Each frame has *m* horizontal Slices made with *k* macroblocks each. This representation corresponds to a 4:2:0 model.

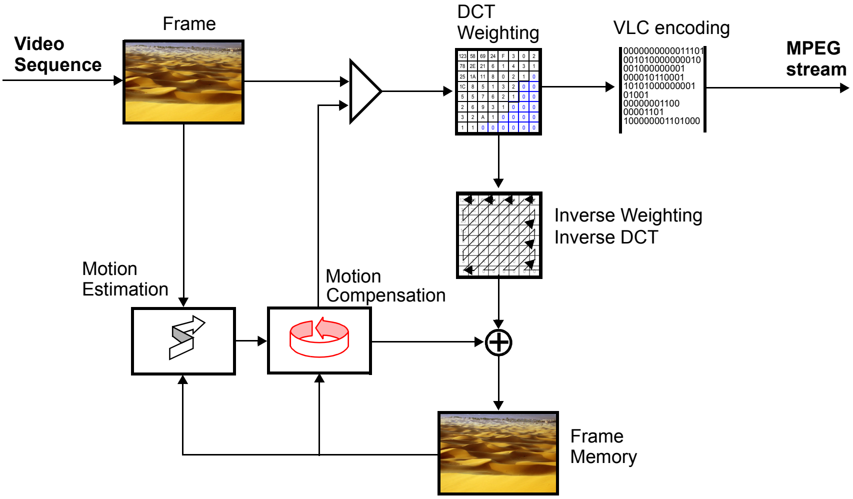
Motion vectors

Motion vectors identify translational changes between the pictures. Block matching search, based on luminance components, identify movements that encoder will calculate and coded into the motion vectors (see Figure 3.16) to predict translations. In the case of P frames motion vectors can refer to previous blocks, and in the case of B frames motion vectors can refer to previous and future frames.

But predictions are imperfect, because compression introduces distortions that result into important differences at the receiver side compared with the original image. To minimize this effect the coder has a decoder to reconstruct images exactly how the receiver would do, and then it is compared with the original. This operation allows the coder to discover the prediction errors and then it can compensate them generating additional information (see Figure 3.17).

3.7.2.3 Codec structure

To implement the above mentioned processes to compress the video streams, MPEG codecs have to combine at several elements such as the DCT procedure to remove the intraframe redundancies, the identification of stationary areas, motion vectors for moving pictures, and VLC coding. In addition the codec also executes the inverse process to identify and compensate compression distortions resulting in an improved video quality (see Figure 3.17).



123	58	69	24	F	3	0	2
78	2E	21	6	1	4	3	1
25	1A	11	8	0	2	1	0
1C	8	5	1	3	2	0	0
5	5	7	6	2	1	0	0
2	6	9	3	1	0	0	0
3	2	A	1	0	0	0	0
1	1	0	0	0	0	0	0

```
0000000000011101
0010100000000010
00100000000001
000010110001
10101000000001
01001
00000001100
00001101
100000001101000
```

Figure 3.17 Simplified model of MPEG-2 encoder.

Buffer Control

After executing several processes and compression algorithms MPEG generates a variable bit rate (VBR) stream, there is an additional issue because the transport is often based on constant bit rate (CBR) channels. A buffer is used to adapt the bit rate between the coder and the transmission channel, then to achieve it and avoid overflows and underflows the buffer level is constantly monitored:

- if the buffer level goes down the coder may increase the size of the DCT output in order to prevent underflow.
- if the buffer level goes high the coder may decrease the size of the DCT output in order to prevent overflow.

The result is a nearly constant bit rate output at the averaged bit rate. At the receiver site the decoder must also have a buffer that will be used to filter the interfamily packet jitter (see Figure 3.28). The size of both buffers should be the same.

3.7.2.4 Type of frames

The simplest approach to the MPEG video is the sequence of several types of frames. The first frame of the group, known as I frame, is fully coded like an independent picture, while the subsequent frames, known as P and B frames, only contain the differences compared to predecessor frames and motion compensation vectors.

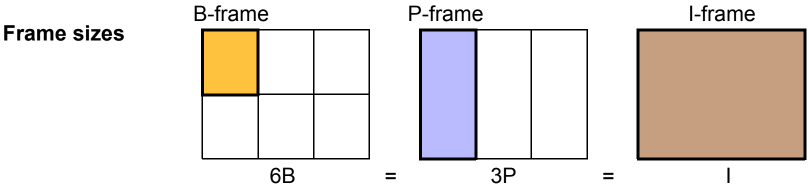


Figure 3.18 B-frames achieve the highest degree of compression, I-frames the lowest.

Lets analyze in a bit more detail how it all works:

- **I-Frames** (Intra Frame) Compression is achieved only removing the spatial redundancy but not the temporal. Therefore I frames do not have references to other frame, and can be reconstructed as it were a single digital picture, totally independent of any other image.
- **P-Frames**: (Prediction Frame) P frames contain only the differences with the previous frame. Blocks can either be coded independly, like I-frame blocks, either use a reference to the nearest I or P frame previously coded and usually incorporating motion compensation.

- **B-Frames:** (Bidirectional-prediction Frame) Each block can either be intra coded or may use backwards or forward references to P and I frames. Forward predictions, or a references to a future frame, require to change the natural frame order before transmission and the receiver must recode again. This operation causes a reordering delay.

A Group of Pictures (GOP) is the sequence of frames embedded between two I-frames. In theory, the number of frames in a GOP is unlimited nevertheless, in practice, typically there are up to fifteen P- and B-frames occurring between two consecutive I-frames. It means that an I-frame could be received, on average, about every 0.4 seconds during the normal video showtime.

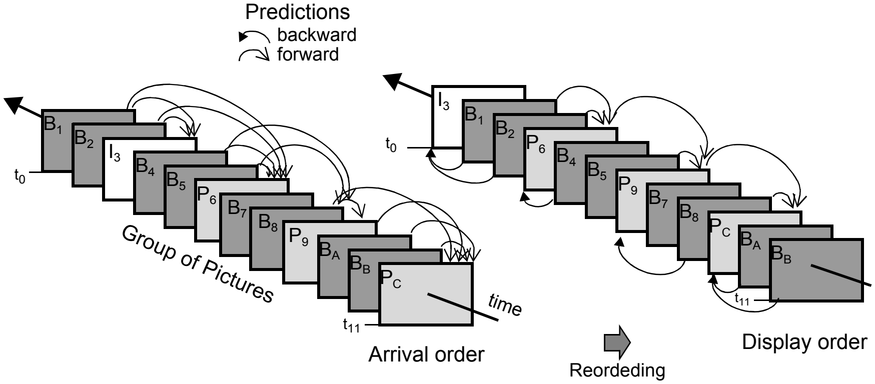


Figure 3.19 Temporal processing sample. A Group of Pictures (GOP) is bounded by I frames with 12/15 frame long. Predictions must be made inside the GOP

Mind about the consequences of a frame lost, and realize that the impact is different depending on the type of frames. If one I-frame is lost, the whole GOP is affected since many of the following frames may refer to it; while the lost of a B frame does not impact to other frames it cannot be used as a reference. The consequences of losing a P-frame can be more similar to lose an I-frame because it can also be used as a reference.

It is also interesting to analyze the impact of frames on the applications:

- In IPTV the STB should wait to receive the first I frame to start displaying images on the customer television.
- A Video compressed only with I frames would allow to stop/forward/backward in every single frame because only I frames are independent. Nevertheless the compression would be minimum.
- A sequence combining I and P frames would achieve better compression but less points to start/stop the video sequence.

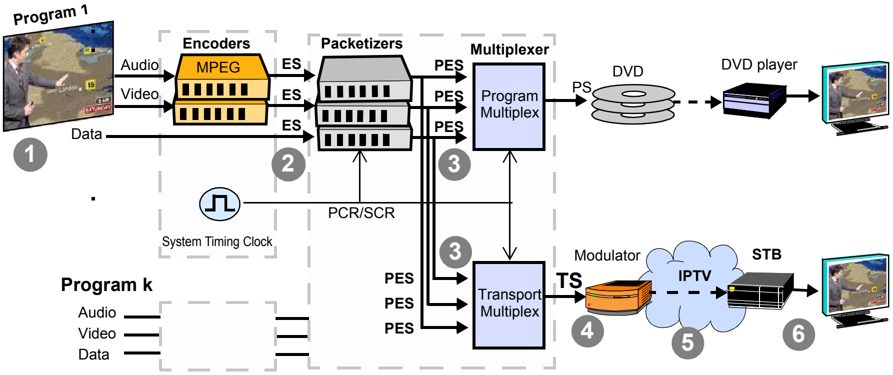
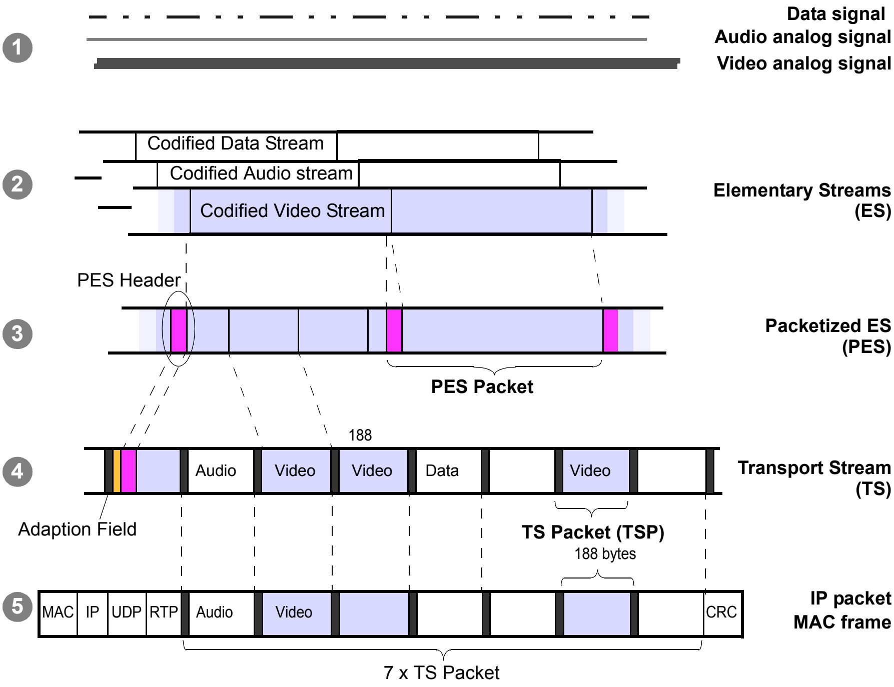
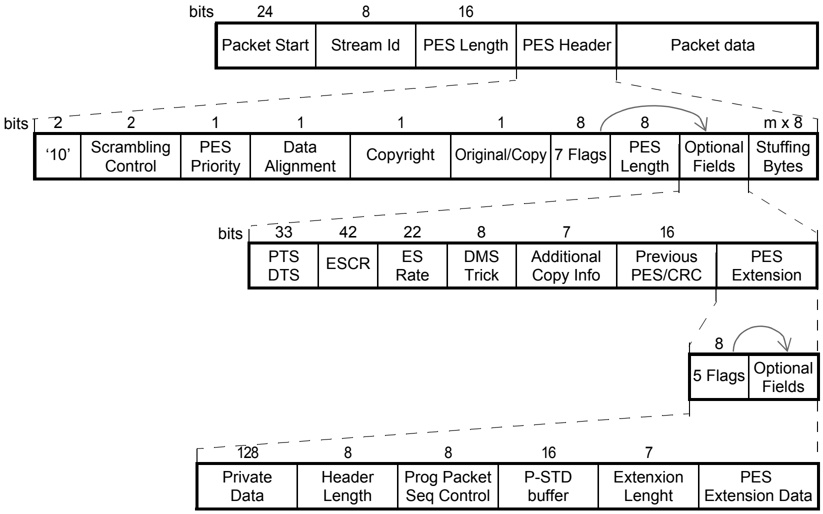


Figure 3.20 (Above) Two type of streams can be generated with the same source signal: Program Stream (PS) and Transport Stream (TS) are two formats of the same structure. The PS is used in CD-ROM and DVD, while TS is intended for transmission over digital Cable, Satellite, or IP centric networks.

Figure 3.21 (Below) Each elementary stream is sent to a processor that creates standard streams called a Packetized Elementary Stream (PES) made up with the original inputs of audio, video and data. The PES are finally segmented into fixed-size Transport Stream Packets (TSP) and put all together to create the Transport Stream (TS)



- A sequence combining all three types achieves max compression but even less points to start/stop the video sequence.



'igure 3.22 Packetized Elementary Stream (PES) Syntax Diagram.

3.7.3 MPEG Stream Generation Scheme

Typically a program has one video stream, one audio stream, and eventually may also have one data stream. Nevertheless other combination are also possible, for instance multilingual programs carry several audio stream and the user choose the idiom. There are also applications that can carry several video streams, a good sample are the multicamera TV programs, where subscribers are invited to select which view want to display at any time, for instance, during a live shows or a sport event.

Whatever the combination is from the TV studio to the customer television, these are the basics steps of an IPTV system (see Figure 3.20):

1. The video camera output is an analog multimedia signal made up with the audio and a video flows.
2. The encoders digitalize the signals and perform the codification by using the MPEG techniques. The output of each individual flow for audio, video and data are the Elementary Streams (ES) (see Figure 3.21)
3. The following step involves packetizers that perform a segmentation of each stream received, adding, the Packetized Elementary Stream Header (PEH) to

identify, synchronize and manage each individual stream (see Table 3.8). The result is a sequence of Packetized Elementary Streams (PES) (see Figure 3.22).

4. The Multiplexer combines all the elementary streams defined for each program in a common Transport Stream (TS) that accepts Transport Streams Packets (TSP) of 188 bytes (see Figure 3.23). Each TSP is made up of a header of 5 bytes and a payload of 183 bytes where are dropped the contents of the PES (see Table 3.9).
5. The TS is split to create the IP/UDP packets that can also be mapped in RTP payload. Generally an IP packet holds up to 7 TSP in the most common case that standard Ethernet frames are used as a transport layer. After streaming the audiovisual information packets are forwarded and distributed across the IP/Carrier-Ethernet network until reaching the destination.
6. Once IP packets are delivered to the STB the inverse operations must be performed to finally obtain the reconstructed audio and video signals ready to be displayed

Two type of streams can be generated with the same source signal: Program Stream (PS) and Transport Stream (TS) that have two different formats despite of the contents is exactly the same. The PS is used in CD-ROM and DVD, while TS is intended to be responsible of the multiplexing and transmission of the digital video and audio elements over a transport network such as Cable, Satellite, or IP centric networks.

Table 3.8
Packetized Elementary Stream Header (PESH).

Byte	*Description*
Packet Start	Equals to ‘0x000001’ identifies the beginning of a packet
Stream Id	specifies the type and number of the elementary stream, i.e.: 110x xxxx - MPEG Audio Stream Number xxxxx 1110 yyyy - MPEG Video Stream Number yyyy
PES Length	Specifies the number of bytes remaining in the packet after this field
Scrambling Cntl	Indicates the scrambling mode of the PES packet payload
PES priority	Indicating the priority of the payload in this PES packet
Data Alignment	Indicates if the header is followed by the video start code or audio syncword
Copyright	Indicates that the material of the associated PES packet payload is protected
Original/Copy	Tells if the contents is an original or is a copy

Table 3.8
Packetized Elementary Stream Header (PESH).

Byte	*Description*
7 Flags	*PTS DTS flags:* Indicates if the PTS fields or/and DTS fields are present
	ESCR flag: Indicates if ESCR base and extension fields are present
	ES rate flag: Indicates if the ES rate field is present
	DSM trick flag: Indicates the presence of an 8-bit trick mode field
	Add Copy flag: Indicates the presence of the additional copy info
	CRC flag: Indicates that a CRC field is present in the PES packet
	PES ext flag: Indicates that an extension field exists in this packet header
Header length	Specifying the total number of bytes
PTS/DTS	Presentation Time Stamp/Decoding Time Stamp: indicates the time of presentation, in the decoder of a audio or video unit of the elementary stream
ESCR	The elementary stream clock reference
ES_rate	Specifies the rate at which the system target decoder receives bytes
Trick Mode	Indicates which trick mode is applied to the video stream: Fast forward i.e. Slow motion, Freeze frame, Fast reverse, Slow reverse
Add Copy Info	Contains private data relating to copyright information
Prev PES CRC	CRC value of the previous packet
5 Flags	*Priv data flag:* Indicates that this packet header contains private data
	Pack header flag: Indicates that an pack header is stored in this packet header
	PPSC flag: Indicates that the Program Packet Sequence Counter is present
	STD buffer flag: Indicates that buffer scale and size parameters are present
	Extension flag: Indicates the presence of the PES extension field length
Private data	Field which contains private data
Header length	This field indicates the length of the header
Prog Seq Count	Optional counter providing functionality similar to a continuity counter
P-STD buffer	Indicates scaling factors to resize the buffer
Extension length	This field which specifies the length the data following

3.7.4 The Transport Stream

Regardless of codec being used (MPEG-2, MPEG-4 or WM9) several are the alternatives for the transport of IPTV payloads over IP:

- *MPEG-2 Transport Stream (TS),* originally designed for MPEG-2 streams it is also used for carrying MPEG-4 or WM9 streams. The TS main job is the multiplexing the digital video and audio stream and to synchronize the receiver. The whole encapsulation is [Codec_Payload + TS + UDP + IP]
- *Real Time Protocol (RTP),* originally developed to carry time sensitive information over IP, can be used to transport and synchronize IPTV streams that are sent separately because it does not have multiplexing capabilities. A good com-

plement is the RTP Control Protocol (RTPCP) which is used to synchronize multiple streams, monitoring the quality and provide feedback. This option that includes is the most more efficient since has less overheads. The whole encapsulation is [Codec_Payload + RTP + UDP + IP]

- A third option, which includes *TS and RTP*, enjoys all the features but it is the less efficient as it increases the overhead. The whole encapsulation is [Codec_Payload + TS + RTP + UDP + IP]

MPEG-2 TS is today probably the most common transport in digital video broadcast (DVB) and also in IPTV. Many of the pioneers on IPTV started with MPEG-2 have decide to keep the same TS, despite the payload actually could be MPEG-4 or WM9, because they are confident with it and already have the mechanisms and the experience to monitor the TS.

3.7.4.1 MPEG-2 Transport Stream

Transport Stream (TS) features include the multiplexing of the audio/video/data streams, the description of the TS structure, indications to decode and display the video, the control of transmission error, and the synchronization of the receiver to correctly display the video.

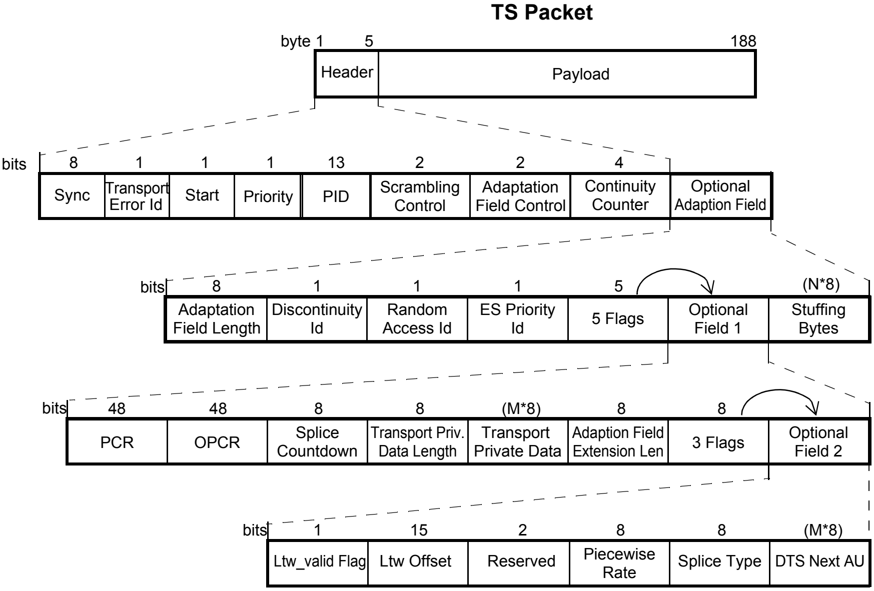


Figure 3.23 Transport Stream Packet Header.

The TS is made up by a continuous sequence of Transport Stream Packets (TSP) that contain the audiovisual information of the programs (see Figure 3.21). The TSP have a header that describe and manage the packet contents (see Figure 3.23). The Packet Identifier (PID) is the first field to decode because it indicates the type information contained in the payload which can be either audiovisual contents or information intended to manage the decoding. Among the fields that manage the decoding there are two essential (see Figure 1.23):

1. Program Specific Information (PSI) to describe the structure of the TS, and
2. Program Clock Reference (PCR) used to synchronize coder and decoder,

Table 3.9
Transport Stream Headers.

Field	*Description*
Sync	Packet synchronization.
Transport Error	Indicates if packet is erroneous.
Start	Indicates if a PES packet header of the start of a table containing program specific information (PSI) is present in the payload.
Priority	0 lower priority - 1 higher priority.
PID	Packet Identifier. Indicates the type of data stored.
Scrambling	Descrambling key to use 00: not scrambled, 10: "even" key, 11: "odd" key.
Adaption	Indicates if whether an adaptation and/or a payload field follows this TS.
Counter	Serial Counter which is used at the decoder to detect lost packets.
Opt. Adaption	Optional field made up a number of subfields
Length	Number of byte in the Adaption Field.
Discontinuity Id.	Indicates time-base or counter discontinuity for the current TS packet
Random Access	Indicates information to aid random access at this point
ES priority	Indicates, among packets with the same PID, the priority of the elementary stream data carried, i.e. if video intra-code slice hold higher priority
5 Flags	PCR flag: adaption field contains PCR
	OPCR flag: adaption field contains OPCR
	Splicing flag: indicates if a splice countdown field is present
	Transport private data flag: adaptation field contains private data bytes
	Extension flag: indicates the presence of an adaptation field extension
Optional Fields 1	Optional field 1 which is made up a number of subfields
PCR	The PCR indicates the intended time of arrival
OPCR	OPCR assists in the reconstruction of a single program TS from another TS
Splice Countdown	A positive value specifies the remaining number of packets of the same PID
	If negative indicates that the packet is the n-th following the splicing point
Transp. Priv. Data	Indicates the number of private data bytes
Priv. Data Byte	Not be specified by ITU-T | ISO/IEC.

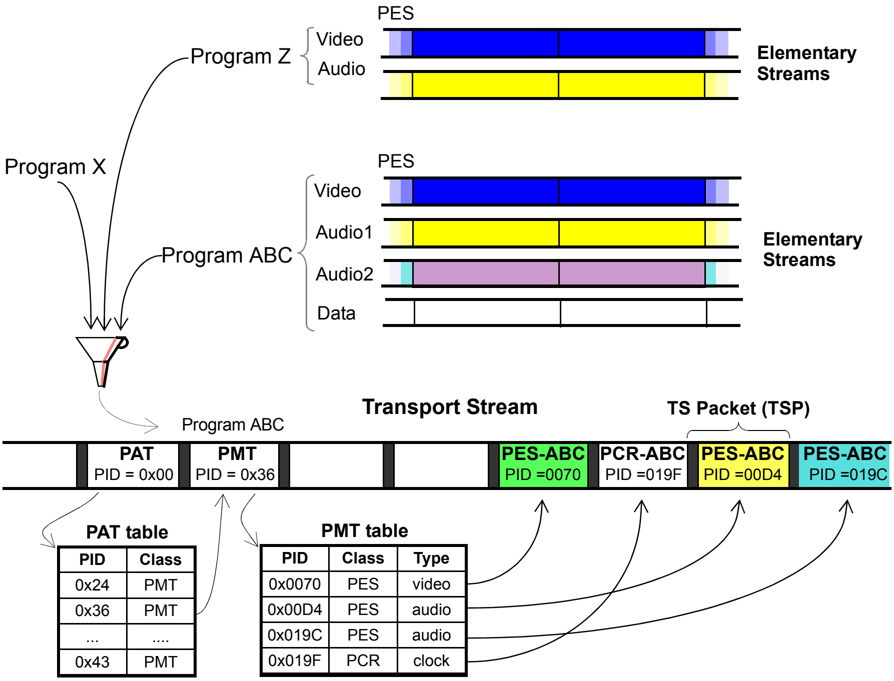
Table 3.9
Transport Stream Headers.

Field	*Description*
Adaption Field Extension Length	Indicates the number of bytes of the extended adaptation field data.
3 Flags	Ltw flag: Indicates the presence of Legal Time Window field. Pwise rate flag: Indicates the presence of the Piecewise Rate field. Seamless splice flag: Splice type and DTS next AU fields are present.
Optional Fields 2	Optional field 1 which is made up a number of subfields.
Ltw valid flag	Indicates that the value of the Legal time window offset shall be valid.
Ltw offset	Information for devices to reconstruct the state of the buffers.
Piecewise rate	Specifies a bit rate which is used to define the end times of the Legal Time Windows of Transport Stream packets of the same PID.
Splice type	If the elementary stream carried in that PID is an audio shall be '0'. If is a video stream indicates the conditions for splicing purposes.
DTS Next AU	Decoding Time Stamp Next Access Unit: In the case of continuous and periodic decoding through this splicing point it indicates the decoding time of the first access unit following the splicing point.

3.7.4.2 Program Specific Information

The Program Specific Information (PSI) are tables embedded in the TS that describe TS structure providing information about the program mapping and relationship between the different transport stream packets (TSP). Some of the PSI have reserved PID to be distinguished quickly by the decoders, while others PID are indicated in the other PSI tables (see Figure 3.24). Here we have some of the most relevant PSI tables:

- *Program Association Table* (PAT), which has a reserved PID=0x00, tells the decoder about programs transported in the TS. The PAT has the pointers to the Program Map Tables (PMT) which in turn identify the video, audio and data TS packets that make up the program.
- *Program Map Table* (PMT), which has a PID indicated in the PAT, identifies and describes all the audio, video and data packet that are part of a each program. Using the PIDs of this packets, provided by the PMT, the receiver can select and combine the different components that are finally decoded and sent to be watched and listened.
- *Conditional Access Table* (CAT), which has a reserved PID=0x01, holds information that is used by the decoder to descramble encrypted programs in services like pay-per-view or private channels. The table list the PID value code that provides the entitlement management messages (EMM).



PAT table

PID	Class
0x24	PMT
0x36	PMT
...	....
0x43	PMT

PMT table

PID	Class	Type
0x0070	PES	video
0x00D4	PES	audio
0x019C	PES	audio
0x019F	PCR	clock

Figure 3.24 The PSI tables describe the structure of the TS providing with the information to retrieve all the contents regarding a program.

- *Transport Stream Description Table* (TSDT), which has a reserved PID=0x02, is a private table implementation defined by MPEG which can be used to provide compatibility with other delivery systems.
- *Network Information Table* (NIT), has a reserved PID=0x10, The contents of this table is private, contains information about the physical parameters of the bearer network such as mappings of user services, channel frequencies, satellite transponder, modulation characteristics, etc.
- *Null Packet*, uses the PID=0x1FFFF, intended for rate stuffing in order to avoid the bottleneck of insufficient resources. The payload is undefined.

The first operation the STB has to do to decode the transport stream is to find the PAT in order to know which programs are available (see Figure 3.24). Then, once identified all the PID of the TS packet that make the program, the decoder has to capture and drop them to be decoded and finally displayed.

3.7.4.3 Timing and Program Clock Reference

Video signals are coded according the System Timing Clock (STC) that runs at 27 MHz which is used to synchronize encoder and decoder. The timing information is included on the transport stream as a time stamp (see Figure 3.20) which is known as Program Clock Reference (PCR). At the receiver site the PCR time stamps are used to synchronize the internal clock which is absolutely fundamental to decode and display the television program.

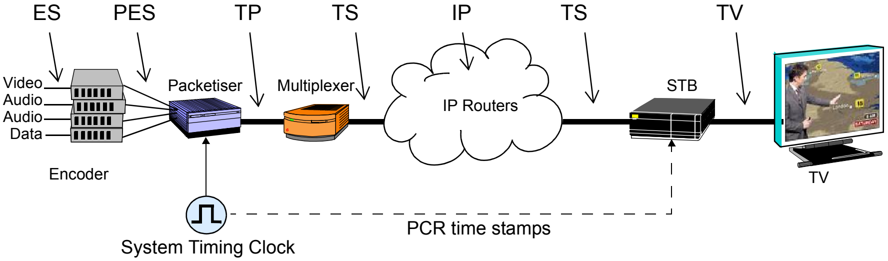


Figure 3.25 IPTV The PCR is used to synchronize encoder and decoder.

3.7.5 Packet distribution and delivery

Converged IP architectures for IPTV do use statistical multiplexing (STDM) to deliver the audiovisual information. In order to distributed IPTV signals across the network until delivering them the subscribers, a IP multicast addressing mode.

3.7.5.1 Broadcast, Multicast, Unicast

Multicasting is the addressing mode to distribute for IPTV to save bandwidth and processing capacity since the unicast and broadcast alternatives do not scale well. Broadcasting every program to all the nodes would be inefficient and would overload unnecessarily links and nodes. Unicast is used for VoD, which works very much how the Internet works, that is establishing an point to point relation between server and subscriber.

But unicast cannot be used for real time IPTV because server and nodes can't deliver well when thousands or even million of subscribers want to watch exactly the same program at the same time. IPTV server forwards a the television datagrams to a multicast enabled router that forwards them to other multicast routers until reaching all hosts associated with the multicast group.

3.7.5.2 The access bottleneck

The limited bandwidth available at the access network means that only IPTV programs that can be received simultaneously. The limit depends on several factors including the video resolution and the access bottleneck that can be partially overpassed using more efficient compression algorithms. In general only a few channels can be delivered simultaneously over the access one or two channels are delivered simultaneously (see Figure 3.7) then the zapping function has to be implemented at the network side after receiving a subscriber requirement to change the channel.

Before receiving the TV signal subscribers must be part of the multicast group to receive or the selected signal using the Internet Group Management Protocol (IGMP). The STB notifies routers their intention to join a multicast group to receive the selected program. When the subscribers want to stop receiving the program a IGMP message is sent to leave the group, then the router no longer forwards datagrams.

3.7.5.3 Multicast and IGMP

The IGMP protocol in combination with multicast networks has been designed to make an efficient use of the available bandwidth at the access that can only accommodate a limited number of IPTV channels simultaneously. The multicast traffic uses a special set of IP addresses in the 224.0.0.0 to 239.255.255.255 range. Each address is associated with a TV channel.

3.8 WINDOWS MEDIA AND VC-1

VC-1 is a video codec specifications that has been recently standardized by the Society of Motion Picture and Television Engineers (SMPTE) based on Microsoft developments. The VC-1 is transport and container independent, it means that can be distributed over MPEG-2 TS and RTP as well.

VC-1 supports three profiles. The Simple and Main profiles have been ready for several years, and existing implementations have supported the creation and playback of video contents. The completion of the Advanced profile represents the final specification that delivers high definition content, either interlaced or progressive, across multiple medium and devices.

3.8.1 VC-1 Profiles and Levels

VC-1 has a number of profile and level combinations to support the encoding of many types of video. The profile determines the codec features that are available, and determines the required decoder complexity..

Table 3.10
VC-1 profiles and levels.

Level	*Profile*	*Bit rate*	*Resolution*
Simple	Low	96 Kbps	176 x 144
	Medium	384 Kbps	240 x 176 352 x 288
Main	Low	2 Mbps	320 x 240
	Medium	10 Mbps	720 x 480) 720 x 576
	High	20 Mbps	1920 x 1080)
Advanced	L0	2 Mbps	352 x 288
	L1	10 Mbps	720 x 480 720 x 576
	L2	20 Mbps	720 x 480 1280 x 720
	L3	45 Mbps	1920 x 1080 1920 x 1080 1280 x 720
	L4	135 Mbps	1920 x 1080 2048 x 1536

3.9 Service provision

Subscribers of IPTV have at least the same expectations, about the quality of the television, than existing TV services. Television is a mature market that have been supplied by many different operators for years. Therefore it is of key importance to match customers expectations from the first day when the service is launched commercially and the subscriber starts to pay, in such a type of mature markets it becomes fundamental a correct plan and a detailed verification of the service quality before offering it commercially. The trial-error-fix strategies do not work and may cause bad consequences for the IPTV business.

The first consideration to take into account is to realize that IPTV is a high bandwidth demanding and also is real time service, which is more sensitivity to the impairments than other services based on voice or data.

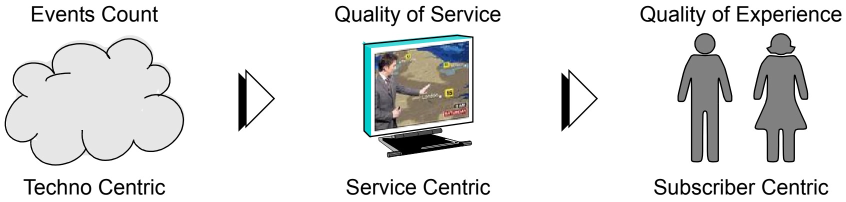


Figure 3.26 Quality of Experience is the human perception regarding how good the service is.

A bad Quality of Experience (QoE) customers suffer has a direct impact on the revenues on the OPEX as increase the calls to the customer support centers that have to fix problem, then often makes necessary to send engineers to the customer premises which is always expensive and defensive activity. More costly could even be the impact on the reputation of the service, we already know some IPTV experiences that after a bad launch that failed in terms of quality and that news quickly were of public domain.

3.9.1 Quality of Experience

Consumers demand high quality for the services but they can't describe the IPTV quality using technological terms. Often the description about the experience is only brought to the service provider's attention when the customer complains in plain terms such as "the image is really bad" or "I can't support such a blurring images" or "we can't zap unless we wait for ten seconds".

Quality of Experience (QoE) is intended to measure how good the service is from the customer's point of view, rather than under the service point of view as the Quality of Service does. QoE is a subscriber centric paradigm (see Figure 3.26) when the IPTV service is degraded producing effect like the following:

- *No Service.* No service at all, frequent service interruptions, wrong formatting, very high latency, authentication problems to watch private programs like pay-per-view.
- *Video degradation*. Frame freezing, blurring, visual noise, loss of color, edge distortion, pixelation, tiling.
- *Audio degradation*. Drop out, bad lips synchronization, voice distortion, bad signal to noise relation, echo.
- *Interaction.* No zapping capability, loss of VoD functions like start/stop/fwd.

Table 3.11
Events that modify the quality of the IPTV services.

Where	*QoS Event*
Head-end & Contents	- Coding distortion - Server overload - Coder impairment - Encoder impairment - Error Indication
Transport Stream	- PCR jitter - Continuity error count - Synchronization error - Interarrival jitter - Error on PSI table - Unacceptable latency
Distribution Network	- Network contention - IP packet loss, jitter, delay - RTP packet loss, jitter - TCP Retransmissions
Transaction	- IGMP latency (IPTV) - RTSP latency (VoD)

Whatever the case is, the QoE is a subjective perception of subscriber that operators has to evaluate measuring an objective combination of parameters. Consequences of a bad service is a miserable video quality which often has same symptoms, independently of the causes, including a combination of frame freezing, pixelation, loss of color, tiling (picture gets broken up into little blocks), chroma noise, or loss of interactivity (see Table 3.11).

3.9.2 Network Impairments

Best effort networks, like the Internet, have difficulties to deliver differential QoS appropriate for each type of application. This fact is a serious inconvenient for all native IP architectures that, in general, need the aid of additional architectonic elements to support IPTV. These features are required to establish admission policies, to control congestion, or simply to manage different traffic priorities. The final objective is to guaranty the QoS required for each application and especially the most sensitive like real-time IPTV. New entrants may have the temptation to set up QoS just over provisioning resources whenever necessary, however we think this strategy is wrong, particularly for the middle and long term could be even more expensive and does not guarantee the result at all.

Quality requirements are different for each application, typically traditional IP data services are happy with best effort networks, but IPTV or VoIP services require more strict control of packet loss, delay, jitter and errors. The ITU-T has published the Rec. ITU-T Y.1541 regarding IP Network QoS control (see Table 3.12). The

Table 3.12
IP Network QoS control (ITU-T Y.1541).

		Upper bound QoS Parameters			
	Applications	*Delay*	*Jitter*	*Loss*	*Error*
Class 0	Real-Time, Jitter Sensitive, High Interaction (VoIP, IPTV, VTC, VoD)	100 ms	50 ms	$1x10^{-3}$	$1x10^{-4}$
Class 1	Real-Time, Jitter Sensitive, Interactive (VoIP, IPTV, VTC, VoD)	400 ms	50 ms	$1x10^{-3}$	$1x10^{-4}$
Class 2	Transaction Data, Highly Interactive (Signalling)	100 ms	Undefined	$1x10^{-3}$	$1x10^{-4}$
Class 3	Transaction Data, Interactive	400 ms	Undefined	$1x10^{-3}$	$1x10^{-4}$
Class 4	Low Loss Only (Short Transactions, Bulk Data, Video Streaming, VoD on local disk)	1 s	Undefined	$1x10^{-3}$	$1x10^{-4}$
Class 5	Best Effort IP Networks	Undefined	Undefined	Undefined	Undefined

achievement of this levels of quality can often be quite challenging. Let's got to analyze in detail most common misfunctions which can be distinguished into two types of impairments:

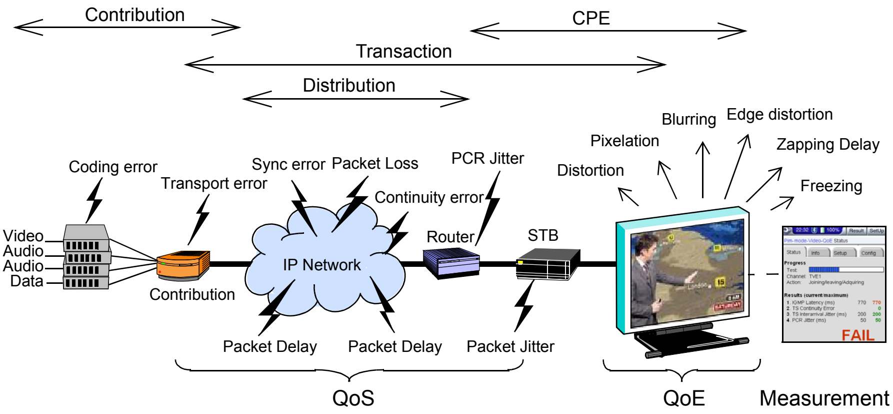
1. **Anomalies**, these are the smaller disagreements that can be observed, such as bit error, but anomalies do not cause interruption of the service. Human intervention is, in principal, not required immediately. However, when the density of anomalies is higher the degradation may affect the quality of the service then the anomaly can be considered a failure.
2. **Failure**s, this event is referred when a fault persists so that the system is unable to perform the expected functionality or even to deliver the service. If the network has been configured as fault tolerant then protection mechanism should be triggered to switch to an alternative power supply, node, path, or subnetwork that would allow to continue delivering the service with a minimum of interruption. Human intervention is often required.

In the particular case of IPTV quality delivered to the subscribers can be affected by many different impairments. Some factors may depend on traffic conditions, this would be the case of the increment of delays, packet lost or errors. Other factors depend on the design and the quality of the network for instance an unacceptable IGMP latency or the encoder impairments.

3.10 SERVICE ASSURANCE

A systematic approach to the IPTV quality begins with a classifications of the service assurance issues in groups, according the place, or the process, where the event occurs:

- **Content Faults.** Includes those events generally produced at the contribution network. Monitoring head-end installations permit the assessment of the video quality and to find out the causes of the misfunction.
- **Network Impairments**, IPTV applications use a transport and distribution network based on the statistical multiplexing therefore traffic conditions may cause contention that produce packet loss, delays and jitter.
- **Transaction Impairments**, the bidirectionality typical of IPTV relay on protocols like IGMP to manage the IPTV channel zapping and RTSP to control de video streaming on VoD. Both may suffer of delay or no functionality at all.
- **Transport Events**, The transport, based on MPEG TS or RTP, is fundamental to distribute, synchronize, and decode the audiovisual streams. Any event at this level can interrupt the ability to decode the programs therein contained.



22:32 100% Result SetUp
Pim-mode-Video-QoE Status
Status Info Setup Config
Progress
Test:
Channel: TVE1
Action: Joining/leaving/Adquiring
Results (current/maximum)
1. IGMP Latency (ms) 770 770
2. TS Continuity Error 0
3. TS Interarrival Jitter (ms) 200 200
4. PCR Jitter (ms) 50 50
FAIL

Figure 3.27 Service assurance for an IPTV service.

3.10.1 Content Faults

This Group embodies those events occurred before the audiovisual signal is streaming to the network and may include server overloads, audio an video coding distortions and error indications.

3.10.1.1 Coding distortion

We have seen a number of coding and compression techniques performed by the encoder at the head end:

- *Interpolative*, like chroma components subsampling,

- *Predictive*, like forward predictions that remove redundancies between frames,
- *Spatial*, like DCT which removes internal redundancies of each frame,
- *Statistical*, like the Hoffman compression based on mathematical codes.

Compression techniques reduce the size of the audiovisual stream but video loss takes place. At the receive site, the decode has to perform the inverse processes to playback the video. However the reconstruction of the original is only one approximation because lossy compressions. This is the coding distortion. The loss factor determines how important the distortion is, and the impact on the quality after the signal has been compressed and decompressed.

3.10.1.2 Error Indicator

This bit is on when the encoders, detect a failure or a corrupted contents of video source. Therefore the presence of this indication is not related with the IP distribution network but revealing that the contribution network has quality issues that require to be addressed.

Table 3.13
IPTV metrics to evaluate the quality

Measurement	*Description*
Video MOS	MOS score in a session
Compression impact	Degradation that can be attributed to the Compression
Frame and bit rates	Video rates in Frames/s and Mbit/s
Bandwidth use (%)	Percentage of bandwidth used at the access
Packet loss (%)	Percentage of packets that were lost
Max burst length	Maximum period of consecutive packet lost
Out of order	Video packets received out of the sequence
Mean jitter	Mean interarrival packet jitter
Max jitter	Maximum interarrival packets jitter
Unavailable seconds	Time without video service or highly degraded
Number of frames I, B, P	Counter of video frames received

3.10.2 Network impairments

3.10.2.1 Bandwidth degradation

IPTV applications are delivered by a packet network that may suffer performance degradation caused by traffic conditions, contention, or events such as a link or a node failure. It is important to identify the causes better during the roll-out of the network o during the trial period. Once the network is in service the traffic must be monitoring continuously at least at the critical points to avoid surprises during the commercial exploitation.

3.10.2.2 Transport Stream Continuity

The mapping of TS Packets (TSP) into Ethernet/IP packets results into frames that contain seven TSPs. Any out of order, lost, or duplication of a single IP packet are errors that will result in a degradation of the video quality. The impact of the error will depend on the nature of the event and the number of packets and the type of frames affected. The lost of I frames is always worse than the lost of B or P frames.

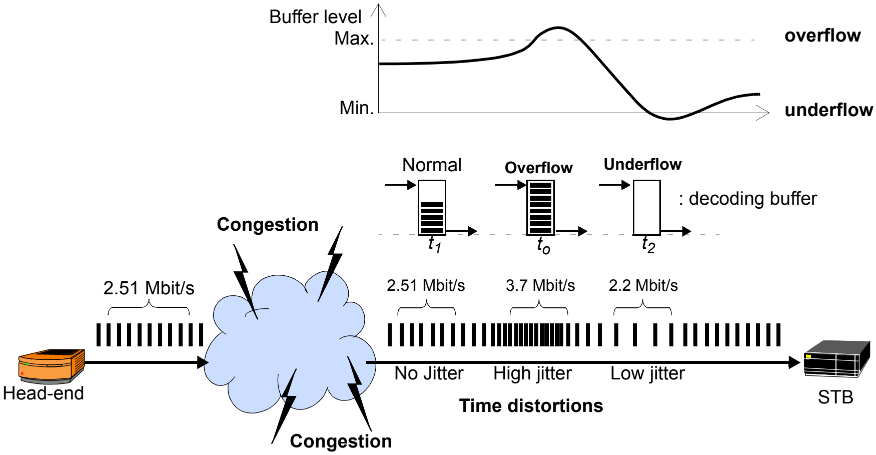


Figure 3.28 Interarrival Packet Jitter is filtered with he decoding buffer but. Higher rates may produce decoder buffer overflow while lower rates me cause buffer underflow. The consequence is, in both cases, a bad quality.

3.10.2.3 Interarrival Packet Jitter

Several are the issues related with the video synchronization. The most common is caused by traffic conditions that produce packet jitter or variation on the delivery of IP packets. This effect generate impairments on recovering the PCR signal which is used to synchronize the 27 MHz receiver clock.

IP packets transporting the audio and video signals are generated with a specific rate at the ingress node of the network. Jitter occurs when packets are delivered in bursts with inter packet gaps shorter or larger than they were generated (see Figure 3.28). Packet Jitter is inherent to the IP networks then receivers use buffers to filter it, the problem is that buffers cannot be too big because they increase the delay. The larger the buffers is the more jitter is filtered, but the packet delay becomes unacceptable then the buffer size is a trade-off between filtering and delay.

There is an obvious limitation of the buffer capacity to filter jitter then further than this edges a buffer overflow or a buffer underflow occurs causing a service degradation (see Figure 3.28). Picks of jitter can cause buffer overflow, while jitter valleys cause buffer underflow. In both cases a temporal but severe degradation of the image occurs, either because there are no frames to decode either because some frames have been dropped.

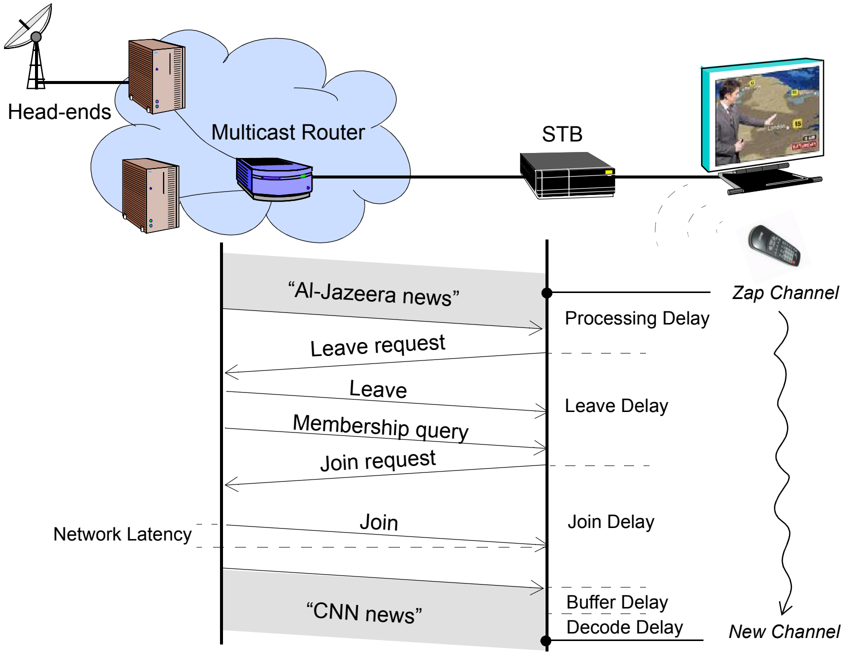


Figure 3.29 Under the customer perception of quality, channel zapping should never take more than one or two seconds.

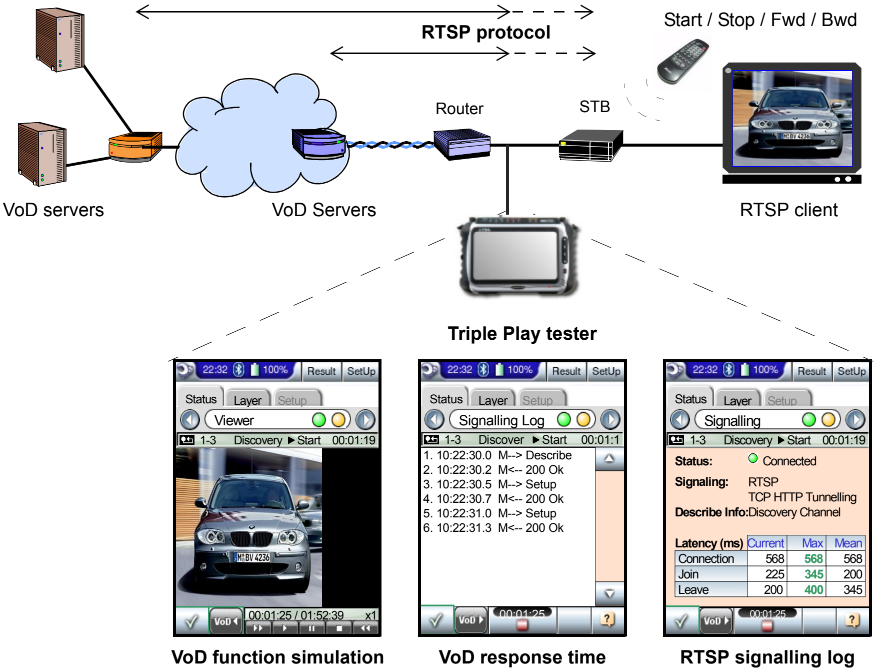
3.10.3 Transaction Impairments

3.10.3.1 Channel Zapping Delay

Channel Zapping Delay is defined as the delay to change programs at the subscriber site. A delay higher than one second will produce a bad perception to the customers that soon will compare with broadcaster's TV ability to zap. The time to perform a channel zap is not only caused by the IGMP protocol execution, there are also other contributors (see Figure 3.29):

- *STB command processing*, is the time it takes between a channel zap is requested and the IGMP leave message is sent.
- *Network Latency*, includes all the transmission latencies due to the network.
- *Leave latency*, is the time it takes to stop the transmission of the old channel
- *Join latency*, is the time it takes to start the reception of the new channel.
- *Buffer delay*, a buffer is necessary to filter the jitter but causes delay.
- *STB delay*, time to find PAT, PMT tables, and the audio, video streams.
- *Decodification delay*, time to find the first I frame and to decode it.

The IGMP delay could be affected by source server, the network traffic, the multicast flow, contention on the servers, etc.



```
1. 10:22:30.0  M--> Describe
2. 10:22:30.2  M<-- 200 Ok
3. 10:22:30.5  M--> Setup
4. 10:22:30.7  M<-- 200 Ok
5. 10:22:31.0  M--> Setup
6. 10:22:31.3  M<-- 200 Ok
```

Status: ● Connected
Signaling: RTSP
TCP HTTP Tunnelling
Describe Info: Discovery Channel

Latency (ms)	Current	Max	Mean
Connection	568	568	568
Join	225	345	200
Leave	200	400	345

Figure 3.30 Verification of the RTSP performance by means of a Triple Play tester.

3.10.3.2 Real Time Streaming Protocol verification

The RTSP is an application-level protocol to manage at the delivery a single or several time-synchronized streams over a continuous media. RTSP provides a framework to enable on-demand delivery of real-time data, such as audio and video. The RTSP verification test should determine the throughput, the individual and the collective response to functions like Play, Pause, Rewind or Forward.

3.10.4 Transport impairments

Defined by the DBV organization, the TR 101 290 is a recommendation to help the operators to set up a digital TV services by means of a set tests that can be applied to MPEG-2 TS interfaces. This transport stream is particularly important because it the most common way to transport, not only MPEG-2 coding, but also MPEG-4 and WM-9.

Tests are grouped into three tables according to their importance for monitoring purposes. The first table lists a basic set of parameters which are considered necessary to ensure that the TS can be decoded. The second table lists additional parameters which are recommended for continuous monitoring. The third table lists optional parameters which could be of interest for some applications.

3.10.4.1 First Table or Priority 1

This category classifies the critical elements necessary for decoding of the MPEG signal. The events can be grouped into three types (see Table 3.14):

- *TS sync loss, PAT and PMT error,* which are events that cause service interruption or severe distortion. For instance if the PAT is missing then the decoder can do nothing, no program is decodable, without a PMT consequences are similar for the corresponding program which cannot be decoded.
- *Continuity Count error*, first indication that the service is degradated, indicate loss of packets, dropped packets, out of order or packet duplications.
- *PID error*, occurs when the data of a referenced PID is lost for a period of time.

Table 3.14
First priority table or basic monitoring necessary for decodability.

Field	*Description*
TS sync loss	Loss of synchronization with consideration of hysteresis parameters
Sync_byte_error	Synchronization byte not equal 0x47

Table 3.14
First priority table or basic monitoring necessary for decodability.

Field	*Description*
PAT Error	PID 0x0000 does not occur at least every 0,5 seconds or a PID 0x0000 does not contain a table_id 0x00
Continuity error	Incorrect packet order, or a packet occurs more than twice, or lost packet
PMT error	Sections with table_id 0x02, do not occur at least every 0,5 seconds on the PID which is referred to in the PAT
PID error	Referred PID does not occur for a user specified period

3.10.4.2 Second table of Priority 2

The following elements should be kept under control to achieve the quality targets (see Table 3.15):

- *CRC error*, indicates corrupted data in the PSI tables.
- *Timing events*, indicates difficulties to recover PCR then synchronization problems are likely.
- *CAT error*, there is an indication to decode a private program but no CAT table has been found to unscramble it..

Table 3.15
Second priority: recommended for continuous or periodic monitoring

Field	*Description*
Transport Error	Transport_error_indicator in the TS-Header is set to "1"
CRC Error	CRC error occurred in CAT, PAT, PMT, NIT, EIT, BAT, SDT or TOT table
PCR Error	PCR discontinuity of more than 100 ms occurring without specific indication. Time interval between two consecutive PCR values more than 40 ms
PCR accur error	PCR accuracy of selected programme is not within ±500 ns
PTS error	PTS repetition period more than 700 ms
CAT error	A packet indicates that data must be unscrambled but no CAT table is found

3.10.4.3 Third table of Priority 3

This is an optional category since many implementations have simplified the structure of the Transport Stream. Then Priority 3 table has a reduced interest for IPTV systems except they have this level of service or require an accurate control of the buffers. The following elements should be kept under control to achieve the quality targets (see Table 3.16):

- *Buffer error*, For this indicator a number of buffers of the MPEG-2 reference decoder are checked whether they would have an underflow or an overflow.

- *Unreferenced PID*, Each non-private program data stream should have its PID listed in the PMTs.
- *SDT error*. The SDT describes the services available. Without the SDT, the STB is unable to give the viewer a list of what services are available.

Table 3.16
Third priority: application dependant monitoring

Error	*Description*
NIT	Id other than 0x40 or 0x41 or 0x72 found on PID = 0x0010
	No section with id 0x40 or 0x41 in PID = 0x0010 for more than 10 seconds
SI repetition	Repetition rate of SI tables outside of specified limits
Buffer error	Overflow of transport buffer, or
	Overflow of multiplexing buffer, or
	Overflow or underflow of elementary stream buffer, or
	Overflow or underflow of main buffer, or
	Overflow of PSI input buffer.
Unreferenced PID	PID not referred to by a PMT within 0,5 seconds
SDT error	SDT TS not present on PID 0x0011 for more than 2 seconds
EIT error	EIT-P/F not present on PID 0x0012 for more than 2 seconds
RST error	Sections with table_id other than 0x71 or 0x72 found on PID 0x0013
TDT error	TDT not present on PID 0x0014 for more than 30 seconds
Empty buffer	Transport buffer not empty at least once per second
Data delay	Delay of data superior to 1 second

3.10.5 Media Delivery Index

The Media Delivery Index (MDI), defined in the RFC 4445, is a measurement for IPTV that gives an indication wether the network is able to transport good quality video independently of the coding scheme. The MDI generally is expressed as two digits XX:YY that refer to on two measurements, the delay factor (DF) and media loss rate (MLR):

- DF is the maximum difference between the arrival and the drain of the TS packets. It is a good indicator about the jitter present in the stream and it is useful to calculate the buffer (or the leaky bucket size) necessary to filter the jitter.
- MLR it is the count of lost and out-of-order TS packets over a period of time. Out-of-order packets are also included because many consumer equipment do not reorder packets.

MLR is considered a good indicator to evaluate the video quality of experience (QoE) using a scientific methodology rather than relaying on subjective human observations like the video MOS. The MDI is often measured at intermediate points of the network to monitor where the issues, that deteriorate the QoE, are located. The methodology is simple, just based on comparing the DF:MLR components between two or more consecutive points and then is possible to quickly identify the nodes or the links that are causing the degradation.

Chapter 4

Signaling

Enabling triple play services involves meeting certain challenges. Some of these challenges are related to the network architecture, others to end-to-end service provisioning. Network-architecture-related issues include finding the right QoS framework and the best technologies to provide bandwidth to subscribers, as well as looking for the optimum switching and routing mechanisms. End-to-end service provisioning involves managing multimedia calls between subscribers and adding intelligence to the network.

Signaling is also a service provisioning issue, and currently there are two approaches to it. The first is the ITU-T approach, described in Recommendation H.323. The solution offered by the IETF is to use the *Session Initiation Protocol* (SIP). These solutions have similar purposes and features, but they follow different design philosophies and are therefore not compatible.

Both H.323 and SIP are essential parts of two different IP multimedia architectures that involve not just one protocol, but many different bits and pieces that can only work properly when they are put together. Some of the protocols that are build on the IP multimedia architecture are the *Real-time Transport Protocol* (RTP), *Real-Time Streaming Protocol* (RTSP) or *Media Gateway Control Protocol* (MGCP) and *Megaco*. Some of them, such as the RTP, form a part of both SIP and H.323. Others, for example the *Session Description Protocol* (SDP) or H.245, are specific to just one of them.

4.1 The Real-Time Transport Protocol

The *Real-time Transport Protocol* (RTP) is a protocol defined by the IETF to carry time-synchronized unicast or multicast data across IP networks. The typical application for RTP is to carry real-time digital audio or video streams, but it can also be used for other applications, such as storage of continuous data.

The RTP protocol provides payload type identification and sequencing, but its most important feature is *payload timestamping*. Packet networks may delay packets of a data stream for a variable amount of time. The RTP is designed to carry timing information with the data. Marking traffic with timestamps makes it possible to re-time the received data with the source timing and with accuracy good enough for many multimedia streaming applications.

Besides unpredictable delays, packets in IP networks may be lost along the transmission path due to congestion or transmission errors. To deal with this situation, a sequence number is included in the RTP. This number allows the receiving end to calculate how many and which packets are missing in the stream.

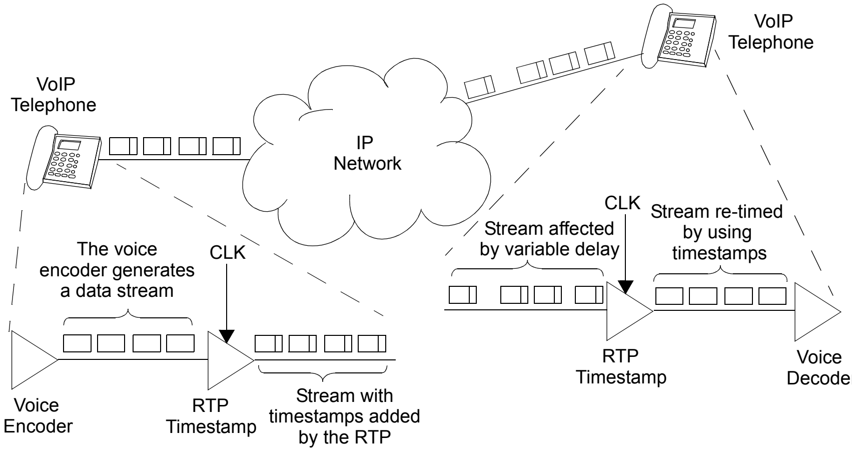

VoIP Telephone
VoIP Telephone
IP Network
The voice encoder generates a data stream
CLK
Voice Encoder
RTP Timestamp
Stream with timestamps added by the RTP
Stream affected by variable delay
CLK
Stream re-timed by using timestamps
RTP Timestamp
Voice Decoder

Figure 4.1 Re-timing of a voice data stream performed by the RTP protocol

Another important aim of the RTP is *payload identification*. There are many alternative ways to transport multimedia streams in packet networks. The RTP provides a common transport framework for all of them. Different payload types may be treated differently at the receiving end. The RTP tells the receiver the type of data that is being transported in the payload and how this data has been encoded.

The RTP is just one component of the IP-based multimedia network framework. It offers some services, such as packet timing and sequencing, but many others are out of the scope of this protocol. For example, the RTP protocol does not offer any mechanisms to guarantee a certain QoS for the data streams it carries. It relies on the QoS mechanisms provided by the network, such as packet marking, shaping and scheduling. Although it is not mandatory, RTP streams are usually delivered by using the *User Datagram Protocol* (UDP) that offers non-connection-oriented transport of IP traffic. The UDP provides low overhead and simplified processing, which is convenient for real-time traffic. The UDP is not a reliable transmission protocol in the sense that it does not offer any error recovery mechanisms. In real-time sessions, there is no time to retransmit lost or corrupted packets. If the data is not tolerant to errors or if the error rate is too high, an error recovery scheme based on *Forward Error Correction* (FEC) must be implemented. Flow control is not supported by UDP either, so some other mechanisms should be used if needed.

4.1.1 Synchronization Sources and Contributing Sources

The RTP streams can be re-timed at the destination, because the generation time of each packet is recorded and delivered with the data. The recorder of the timing information is identified by a 32-bit number and sent with the data as well. This identifier is called *Synchronization Source* (SSRC). The same transmitter may generate multiple RTP streams with different SSRCs. For example, two RTP streams with two different SSRCs may be transmitted during a videoconference. One of them for the video, and the other one for the audio stream. Each SSRC, even when coming from the same source, has dedicated sequencing and timestamping.

The simplest case of an RTP stream is communication between two parties; a transmitter and a receiver. A VoIP telephone call to a single person is a good example. In this case, the transmitter encodes the data to be transmitted, and generates an RTP stream. The data is sent directly to the receiver. The transmitter identifies itself with its own IP address, but also with a randomly chosen 32-bit number: the SSRC.

Note that the SSRC is chosen randomly by the transmitters. This would probably not cause a collision, but the parties involved in the transmission must be able to identify and solve this situation.

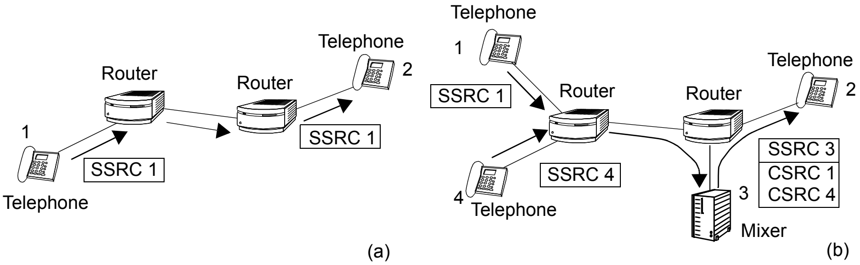


Figure 4.2 Examples of RTP streams: (a) Simple RTP session with one transmitter and one receiver. The transmitter identifies itself as the synchronization source for the RTP stream. (b) Multiparty RTP stream. The transmitters deliver the traffic to an intermediate host, a mixer. The streams are combined, re-timed and delivered to the destination by the mixer. The mixer identifies itself as the synchronization source for the new RTP stream, but the original transmitters are still identified as contributing sources.

Often, RTP applications involve more than one receiver or more than one transmitter. Distributing TV signals or multiparty VoIP telephone calls are two important examples of this. When the multi-transmitter situation arises, it might be necessary to combine the streams from every transmitter into a single RTP stream before delivery. There is a special device in charge of this task called mixer.

If a mixer is used, RTP streams are not sent directly to the destination, they are delivered first to the mixer. Thus, the destination IP address of the RTP packets is set to the IP address of the mixer. The transmitters identify themselves by their IP address and SSRC numbers. Once the packet arrives to the mixer, it is processed and a single RTP stream is generated from the multiple input streams. The mixer sends the new RTP data to the receiver. To do this, it is necessary to set the destination address of the new packets to the IP address of the receiver. The mixer identifies itself by setting the source address of the RTP packets to its own IP address, and setting its own SSRC. The original sources of the RTP stream processed by the mixer are listed in the RTP packets as *Contributing Sources* (CSRC). The number of CSRCs in the RTP stream is limited to 15. When the transmission involves more than 15 transmitters, only some of them can be identified.

4.1.2 Translators and Mixers

Translators and mixers are intermediate systems that work at the RTP level. They are in charge of protocol and format translations.

An RTP framework involving only end-user network equipment would be more simple, but some situations arise in practice that makes it necessary to use mixers and translators to accommodate the network's resources and data format for different receivers.

Mixers and translators are similar, but the former can do more than the latter:

- *Translators* are intermediate network systems that work at the RTP level. They usually change the format of the incoming RTP packets and forward them to their final destination with their SSRC identifier untouched. The operations performed by translators include downsampling of RTP data for low bandwidth networks, changing the encoding of video or audio streams, and tunneling of multicast data across firewalls.
- *Mixers* can potentially do the same tasks as translators, but additionally they can combine streams from different sources into one single outgoing RTP stream. Multi-party telephone call is an application that is easy to understand, but some other applications are possible as well. When a mixer combines data from different sources, it cannot choose the SSRC identifier from any of them, and thus puts its own SSRC in the forwarded data. The identity of the original generators is preserved by adding it to the CSRC list. Mixers contribute to save

bandwidth in the network, because the bit rate of the outgoing traffic is similar to the bit rate of a single source, and smaller than the sum of all the sources.

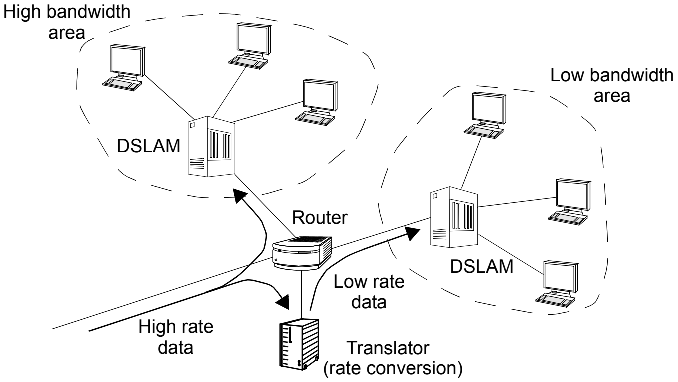


Figure 4.3 One application for translators is rate conversion. They can down-convert the rate of RTP data to be routed to users with low-rate access to the service. This approach makes it possible for the users to take the maximum profit out of their connections.

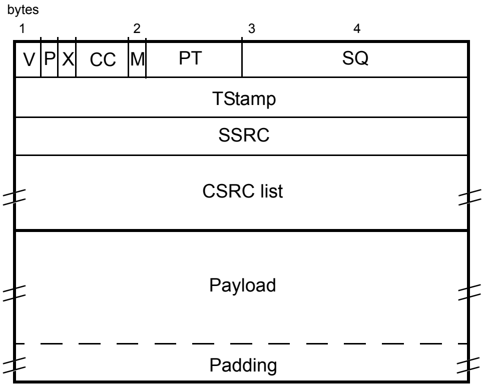
4.1.3 The RTP Packet

The RTP features are only shown in the packet format of the protocol. RTP packets provide a way to define the payload type, use timestamps, sequence numbers, SSRC, CSRCs and so on. The fields of an RTP packet header are the following:

- *Version (V)*: Gives information on the version of the RTP. The current version is 2.
- *Padding (P)*: This bit is set to 1 to tell the receiver that the packet contains one or more additional padding bytes at the end that do not carry information. The number of bytes to be discarded is carried in the last padding byte.
- *Extension (X)*: This field is set to 1 to tell the receiver that the optional RTP header extension is enabled. Header extensions may be implemented by proprietary versions of the RTP that require additional information to be carried in the RTP header.
- *CSRC Count (CC)*: Number of CSRC identifiers for the RTP packet. All CSRC identifiers are listed in the CSRC list field of the RTP header.
- *Marker (M)*: The actual meaning of this bit depends on the profile. It can be used to mark boundaries of higher-level frames, or to signal extra payload types. It may remain unused in some applications.
- *Payload Type (PT)*: PT specifies the payload format and media type carried in the RTP stream. It is used by the receiver to choose the right decoder. Mapping

different encoders and payload formats to bit codes depends on the profile. However, there are default mappings for many common media formats.

- *Sequence number (SQ)*: SQ is a number chosen randomly by the receiver when starting to transmit an RTP stream. This number is incremented by one for each packet sent. This makes it possible for the receiver to detect packet loss.
- *Timestamp (TStamp)*: TStamp is a number that represents the sampling instant of the first byte in the RTP packet payload. It allows the receiver to re-time the incoming data according to the clock used to generate the stream. The frequency of this clock may be different for different payload types, and it is identified in the profile or in the payload format specification. The initial value for this field is chosen randomly.
- *Synchronization Source (SSRC)*: Randomly chosen number, identifying the originator of the RTP data stream.
- *Contributing Source list (CSRC list)*: List of contributing sources for the packet payload. The number of items on this list is signaled by the CC header field. The list is limited to 15 items, and a list of 0 items is possible.



V: Version (2 bits)
P: Padding (1 bit)
X: Extension (1 bit)
CC: CSRC Count (4 bits)
M: Marker (1 bit)
PT: Payload Type (7 bits)
SQ: SeQuence number (16 bits)
TStamp: TimeStamp (32 bits)
SSRC: Synchronization SouRCe (32 bits)
CSRC list: Contributing SouRCe list
(32 bits x number of items)

Figure 4.4 RTP packet format

Table 4.1
Payload types for RTP defined at RFC 3551

PT	*Name*	*Encoder*	*Media*
0	PCMU	PCM with μ-law scaling as per ITU-T G.711	Audio
3	GSM	GSM standard ETS 300 961	Audio
4	G723	ITU-T Recommendation G.723.1	Audio
5	DVI4	IMA ADPCM wave type, 8,000 Hz clock	Audio
6	DVI4	IMA ADPCM wave type, 16,000 Hz clock	Audio
7	LPC	Linear Predictive Encoding contributed by R. Frederick	Audio

Table 4.1
Payload types for RTP defined at RFC 3551

PT	*Name*	*Encoder*	*Media*
8	PCMA	PCM with A-law scaling as per ITU-T G.711	Audio
9	G722	ITU-T Recommendation G.722	Audio
10	L16	Uncompressed audio, 16 bits, 44100 Hz and 2 channels	Audio
11	L16	Uncompressed audio, 16 bits, 44100 Hz and 1 channel	Audio
12	QCELP	EIA / TIA standard IS-733	Audio
13	CN	Comfort Noise as per RFC 3389	Audio
14	MPA	ISO/IEC 11172-3 and 13818-3 MPEG-1 / MPEG-2 audio	Audio
15	G728	ITU-T Recommendation G.728	Audio
16	DVI4	IMA ADPCM wave type, 11,025 Hz clock	Audio
17	DVI4	IMA ADPCM wave type, 22,050 Hz clock	Audio
18	G729	ITU-T Recommendation G.729	Audio
dynamic	G726-40	ITU-T Recommendation G.726, 40 kb/s ADPCM	Audio
dynamic	G726-32	ITU-T Recommendation G.726, 32 kb/s ADPCM	Audio
dynamic	G726-24	ITU-T Recommendation G.726, 24 kb/s ADPCM	Audio
dynamic	G726-16	ITU-T Recommendation G.726, 16 kb/s ADPCM	Audio
dynamic	G729D	ITU-T Recommendation G.729 annex D	Audio
dynamic	G729E	ITU-T Recommendation G.729 annex E	Audio
dynamic	GSM-EFR	GSM standard ETS 300 726	Audio
dynamic	L8	Uncompressed audio, 8 bits samples	Audio
dynamic	RED	REDundant audio payload as per RFC 2198	Audio
dynamic	VDVI	Variable rate version of DVI4	Audio
25	CelB	Sun Microsystems CELL-B encoding	Video
26	JPEG	ISO Standards 10918-1 and 10918-2	Video
28	nv	“nv” encoding, version 4 developed at Xerox PARC	Video
31	H261	ITU-T Recommendation H.261	Video
32	MPV	ISO/IEC 11172-3 and 13818-3 MPEG-1 / MPEG-2 video	Video
33	MP2T	MPEG-2 transport streams for either audio or video	Audio / Video
34	H263	ITU-T Recommendation H.263 of 1996	Video
dynamic	H263-1998	ITU-T Recommendation H.263 of 1996	Video

4.1.4 Stream Multiplexing

There are different fields in the RTP packet or in the underlying protocols that can be used for stream multiplexing. The SSRC or the PT may be used, as well as the IP address or the transport layer port. Some of the available options are better than others, and some of them must be avoided.

Using the PT field of the RTP packet has several major inconveniences and must be avoided:

- The format of one or more payloads and their associated PT field values may change during time for the same stream, so, it is not a very good idea to use this field to identify the stream.
- Every payload type may require specific sequencing and timestamping and therefore dedicated SSRCs.
- Mixers may have problems to combine incompatible streams. Care must be taken to avoid carrying incompatible payloads under the same SSRC.

But using just the SSRC for multiplexing makes it difficult to take profit of some customized IP and UDP layer processing.

- It would be difficult to use different network paths or receive custom QoS treatment for different media types.
- Networking software makes use of transport layer ports to listen to the network. Two or more processes cannot share the same port, and therefore it is not possible to use customized processes to decode specific media if only SSRC but not the transport layer port is used for multiplexing.

For all these reasons, different RTP media types, for example the audio and video in a videoconference, are normally transmitted with different transport layer ports and / or different IP addresses.

4.1.5 Security

There is a clear need for security in multimedia communications over packet networks. Security in communications is a term that includes different issues:

- *Confidentiality* means that only the intended receiver of the data stream will be able to decode and understand the message. Confidentiality is provided by encrypting the data.
- *Authentication* refers to the ability of the receiver of a data stream to prove the identity of the generating entity. Digital signatures are used for this.
- *Integrity* is achieved when the receiver is able to prove that the message has not been modified by a third party before reception. Here, message-dependent signatures are used that only the transmitter is able to generate.

The RTP protocol supports confidentiality, but authentication and message integrity cannot be provided without a key management and distribution framework that is not natively supported by RTP.

Confidentiality is provided by means of the *Data Encryption Standard* (DES) algorithm mode. If more options or stronger encryption are needed, it is possible to implement encryption in other protocol layers, for example by means *IP security* (IPsec).

4.2 The Real-Time Control Protocol

The *Real-Time Control Protocol* (RTCP) complements the RTP in the following ways:

1. It gives feedback on the quality of transmission. This is, in fact, the primary role of the RTCP. This function is related to the flow and congestion control performed by other transport layer protocols.
2. It carries persistent transport-level identifiers for RTP sources. The SSRC may not be enough for some applications, because it can change when a collision occurs or if the end user equipment is restarted. The RCTP can supply complementary source information.
3. It provides minimum session control. It does not support complex parameter negotiation between the communicating parties, nor membership control. If these advanced features are needed, a higher-level session control protocol must be used.

These functions are mandatory for multicast applications, and recommended for all applications. To accomplish these functions, the RTCP packets are multiplexed with the RTP stream. Usually, ports are used for multiplexing; the RTP uses even-numbered ports, and the RTCP uses the contiguous higher odd port number when it is delivered over UDP. The bit rate of RTCP is limited to maintain the performance of the RTP streams that carry user data. For a session, the RTCP bandwidth is usually no more than 5 %.

4.2.1 RTCP Packet Types and Formats

The RTCP packet format depends on the purpose of the packet. The header always contains a 2-bit version field that must match the version field of the RTP header, a padding indicator bit, an 8-bit PT field and a 16-bit length field. Other fields may be present depending on the RTCP packet type.

The receiver knows the RTCP packet type by means of the PT field that is always the second byte of the RTCP packet header. This field may also be used to differentiate RTP and RTCP protocols, because the PT identifiers used by the RTCP are reserved. There are five RTCP packet types: *Sender Report* (SR), *Receiver Report* (RR), *Source Description* (SDES), *end of participation* (BYE) and *Application-specific packet* (APP).

Table 4.2
RTCP Packet Types

Type	*PT id.*	*Purpose*
SR	200	Sender report: quality statistics from active senders
RR	201	Receiver report: quality statistics from receivers that are not senders
SDES	202	Source description items: information on the transmitter
BYE	203	Indicates end of participation
APP	204	Application-specific functions

It is possible for translators and mixers to process and modify RTCP information when it passes across them. For example, several RTCP packets may be combined into a single packet in order to decrease the overhead / payload ratio.

4.2.2 Quality of Service Monitoring

The quality of a packet-based multimedia service can be evaluated with some performance parameters; the most important being delay, jitter and packet loss. The RTCP enables the RTP receiver to generate periodic reports that provide the RTP transmitter or intermediate probes with quality feedback on these three performance parameters.

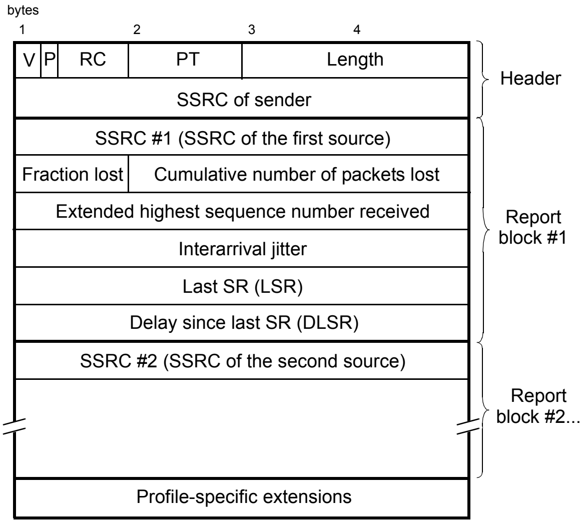

bytes
1
2
3
4
V
P
RC
PT
Length
SSRC of sender
Header
SSRC #1 (SSRC of the first source)
Fraction lost
Cumulative number of packets lost
Extended highest sequence number received
Interarrival jitter
Last SR (LSR)
Delay since last SR (DLSR)
Report block #1
SSRC #2 (SSRC of the second source)
Report block #2...
Profile-specific extensions

Figure 4.5 RCTP RR packet format

Statistics are calculated by the communicating parties with the help of specific RTCP fields containing timestamps and packet counts. The reports are generated periodically. If a particular party has transmitted one or more RTP packets since the last report, it will generate a report called SR. Parties that have not sent any RTP packets since the last quality report generate an RR report.

The RR includes information on the sources (identified by their SSRC) from which the receiver has received RTP data packets since the latest report. The number of sources included in the report is given by the *Reception Report Count* (RC) field of the RTCP packet header. This field has 5 bits, and therefore the maximum number of sources that can be identified ranges from 0 to 31. The SR contains similar information as the RR, but the PT field is different, and a group of fields includes specific statistics on the RTP transmitter.

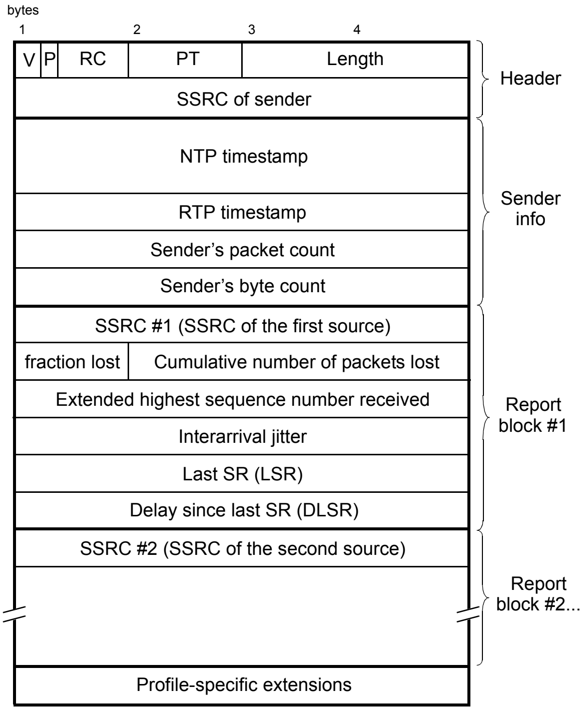

bytes
1
2
3
4
V
P
RC
PT
Length
SSRC of sender
Header
NTP timestamp
RTP timestamp
Sender's packet count
Sender's byte count
Sender info
SSRC #1 (SSRC of the first source)
fraction lost
Cumulative number of packets lost
Extended highest sequence number received
Interarrival jitter
Last SR (LSR)
Delay since last SR (DLSR)
Report block #1
SSRC #2 (SSRC of the second source)
Report block #2...
Profile-specific extensions

Figure 4.6 RCTP SR packet format

4.2.2.1 Delay Statistics

The RTCP enables the transmitters to get an estimation of the *Round Trip Delay* (RTD), and makes it possible for the receivers to estimate one-way delay.

One-way delay can be calculated as the difference between the originating packet timestamp and the arrival time of the packet. The originating timestamp is recorded by the transmitter and sent to the receiver in a 64-bit field of the SR. The arrival time of the packet is recorded to calculate one-way delay, but it is also sent back to the transmitter within the 32-bit LSR field of the following RR. This makes it possible to calculate the RTD.

The difference between the time the RR arrives to the transmitter and the LSR field of this packet is not the RTD, because it includes the time between receiving the SR and generating the RR. This time does not depend on the transmission performance, but relies on the RTCP protocol itself, which is why it must not be taken into account when calculating the RTD (see Figure 4.7).

For delay estimations, the transmitter and receiver clocks have to be synchronized. The *Network Time Protocol* (NTP) is used for this purpose. The NTP is a popular protocol used to synchronize the internal time of computers and routers connected to Internet with timing sources that provide an accurate *Universal Time Coordinated* (UTC) time. The problem is that the accuracy of NTP-based timing is quite unpredictable and ranges from tens of milliseconds to less than one microsecond.

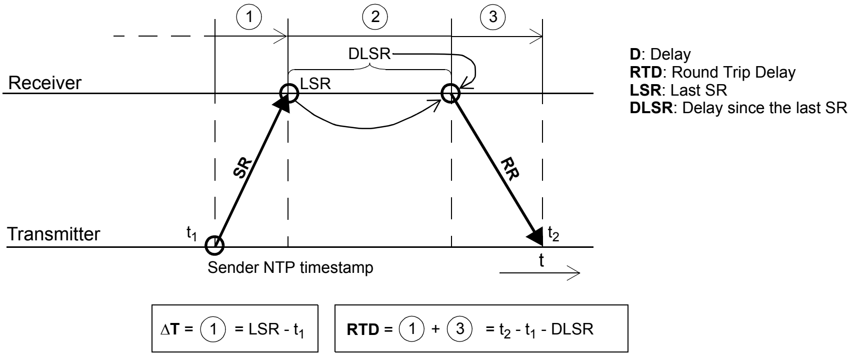


Figure 4.7 Timestamps sent with SR and RR, and estimation of one-way delay and round trip delay

Besides NTP timestamps, an RTP timestamp is also transported in the SR. The RTP timestamp is the same as the NTP timestamp, but measured in RTP units. It is used to synchronize different media of the same source, or to estimate the clock frequency of the source.

4.2.2.2 Jitter Statistics

The receivers provide an estimate of the RTP packet interarrival jitter that is recorded in the corresponding 32-bit field of the RR. Interarrival jitter is defined as the average difference of the delay between two consecutive RTP packets.

If ΔT_i is the one-way delay of a packet (i), the delay difference between two packets is calculated as shown:

$$\Delta\Delta T_{ij} = \Delta T_j - \Delta T_i \qquad \textit{eq. 4.1}$$

Interarrival jitter, J , is related to the mean value of $\Delta\Delta T_{ij}$. An estimate of interarrival jitter at a given time (i) is given by:

$$J_i = J_{i-1} + \frac{|\Delta\Delta T_{i-1,i}| - J_{i-1}}{16} \qquad \textit{eq. 4.2}$$

4.2.2.3 Packet Loss Statistics

There are several fields in the SR and RR that contain information that is useful for estimating packet loss. The RTP data source send the following parameters to the destination:

- The number of packets transmitted since the beginning of the RTP session. This parameter is transported in the *Sender's Packet Count* field of the SR.
- The number of payload bytes transmitted since the beginning of the RTP session. This figure is sent in the *Sender's Byte Count* field of the SR.

An RTP receiver can estimate transmission performance by using the RTP sequence numbers and the SR data received in RTCP packets. Regarding packet loss, the RTP receiver sends the following information back to the transmitter:

- The highest sequence number received in an RTP packet. The difference between the values of this parameter at two different points in time is the expected number of RTP packets in the time interval. The expected number of packets since the beginning of the session is the highest sequence number minus the randomly chosen initial sequence number. This parameter is sent to the transmitter in the *Extended Highest Sequence Number Received* field of the SR or RR.
- The cumulative number of packets lost since the beginning of the session. This parameter is calculated as the number of received packets minus the expected number of packets. This parameter is included in the *Cumulative Number of Packets Lost* field of the SR or RR.

- A fraction of RTP data packets lost since the previous SR or RR. This parameter is calculated as the estimated number of packets lost since the last report, divided by the number of packets expected. This data is delivered to the RTP transmitter within the *Fraction Lost* field of the SR or RR.

4.2.2.4 Derived Statistics

Delay, jitter and packet loss are important parameters when evaluating the performance of RTP sessions, but there are other possibilities to do this as well. New performance parameters can be derived from the existing ones.

The cumulative number of packets lost at two different points in time can be used to estimate the number of packets lost during that interval. This value is divided by the expected number of packets in that interval, and the result is the fraction of lost packets. This ratio will only match the value of the *Fraction Lost* field if it is calculated over two consecutive RTCP packets.

By relating the primary performance parameters, the following values can be obtained as well: the rate of lost packets per second (packets/s), the number of received packets, average payload data rate (bytes/s), the average packet rate (packets/s), the average payload size (bytes/packet), etc.

4.2.3 Source Description

The SDES RCTP packets give a description of the RTP source. The SDES packet has a similar structure as the other RCTP packets. It has a 32-bit header followed by source description items for one or more RTP sources identified by their SSRC or CSRC. The terminal RTP equipment generates SDES packets with just one SSRC, but a mixer may contain information for many contributing sources, and the SDES packet may contain many chunks identified by their CSRC. The number of chunks is always identified in the *Source Count* (SC) field of the header.

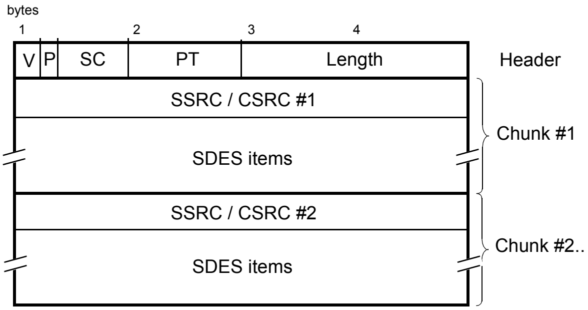


Figure 4.8 RCTP SDES packet format

All SDES items have similar format:

- an 8-bit identifier showing the type of information included in the SDES item
- an 8-bit field that includes the length of the SDES item
- a variable-length field that contains some kind of information about the source

The information transported in SDES packets is miscellaneous. The most important SDES item is the *Canonical end-point identifier* or CNAME. The CNAME is a unique identifier in the RTP session that binds a data source with an alphanumeric identifier. This makes it possible for the destination to identify the source.

Note that the SSRC has a similar objective, except that the SSRC can change during the session if a collision with other SSRC is detected or if the user equipment is restarted. This will not happen with the CNAME. Another interesting property of the CNAME is that it identifies the parties of an RTP session better than the synchronization sources. It could be useful to relate different media coming from the same transmitter but with different SSRCs.

The CNAME has a "user@host" format, but it is not an e-mail address. The "user" is normally the login name of a participant in the RTP session and the "host" identifies the source host. It is normally either the fully qualified domain name of the host, or an ASCII representation of its IP address in dotted decimal format.

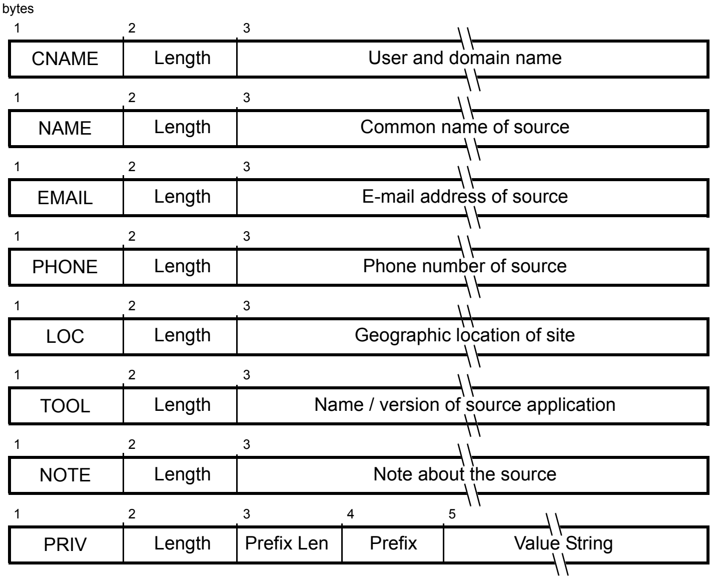

bytes
1 2 3
CNAME | Length | User and domain name
1 2 3
NAME | Length | Common name of source
1 2 3
EMAIL | Length | E-mail address of source
1 2 3
PHONE | Length | Phone number of source
1 2 3
LOC | Length | Geographic location of site
1 2 3
TOOL | Length | Name / version of source application
1 2 3
NOTE | Length | Note about the source
1 2 3 4 5
PRIV | Length | Prefix Len | Prefix | Value String

Figure 4.9 Currently defined SDES items

There are other SDES item types that can be useful for RTP applications:

- NAME: An alphanumeric string that identifies the source, normally displayed. For example, it could include the names of the participants in an RTP application. Normally, it doesn't need to be unique for all participants. For example "John Smith, Albedo Telecom.".
- EMAIL: The e-mail address of the participants in an RTP session formatted as specified in RFC 822. A valid value for an EMAIL field would be "info.telecom@albedo.biz".
- PHONE: The telephone number of the participants in an RTP session, with the plus sign replacing the international access code. For example "+44 1628 503500".
- LOC: An alphanumeric string that identifies the geographical location of the participants in the RTP session. The level of detail and format may depend on the application. For example "Lower Cookham Road, Maidenhead".
- TOOL: A string including the name and in some cases the version of the RTP application. For example "Albedo Videocom 1.12".
- NOTE: This SDES item carries transient messages describing the current state of the source. For example "out for lunch" or "busy".
- PRIV: Application-specific or experimental SDES messages. This SDES item includes a prefix of variable length that may contain a description of the value of this field.

4.2.4 Session Management: The BYE Packet

The RTCP is not a session management protocol. However, it can provide minimum session control in very simple environments. Some SDES messages may be useful to interchange session information between the parties of an RTP communication. Another useful feature is the *Goodbye Packet* (BYE), which shows that one or more sources are no longer available.

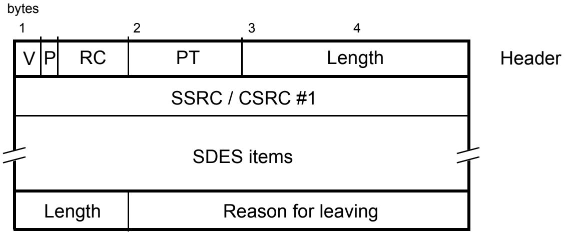


Figure 4.10 BYE packet format

4.3 The Session Initiation Protocol

The *Session Initiation Protocol* (SIP) is an application layer protocol whose main purpose is to establish, modify and tear down sessions between users in IP networks. The first version of SIP was released by the IETF in 1999, and it is expected to have a similar importance as HTTP for web applications and SMTP for e-mail. The SIP protocol can be used with any kind of sessions, including telephony, fax, videoconferencing and instant messaging, but an important goal of this protocol is to be able to setup real-time communication sessions across the Internet.

The purpose of SIP can be compared with the Q.931 or *ISDN User Part* (ISUP) protocols, but while Q.931 and ISUP are restricted to signaling in the PSTN, SIP is designed for Internet. A major difference between SIP and Q.931 or ISUP is that SIP does not make resource reservation. It only finds users in the network and sets up sessions with them.

Besides session establishment and tear-down, the SIP is able to locate users identified by a *Uniform Resource Identifier* (URI). The protocol provides specific SIP URIs for this purpose, with a “sip:user@domain” structure similar to e-mail addresses. Additionally, secure URIs, called SIPS URIs are provided. The SIPS URIs have a “sips:user@domain” structure. For example, “sip:alice@albedo.com” and “sips:7545@pc25.securedomain.org” are valid SIP or SIPS URIs. If a domain name is not available, IP addresses can be used instead. For example, “sip:mike@10.0.0.15” is a valid SIP URI. It is possible to establish sessions with remote users by means of other URIs different to the SIP or SIPS URIs as well. For example, the *tel URL* (RFC 2806), to call traditional telephones, is compatible with SIP.

In SIP, Internet telephones, conferencing software and other kinds of end-user equipment are called user agents. User agents are very different to users. URIs identify users rather than user agents. Users are allowed to keep the same identifier (their URI) even if they change their connection points to the network. This is again similar to what happens with e-mail addresses. A particular user agent can be attached to several user agent equipment, or several users may be allowed to use the same user agent equipment. The basic purpose of this approach is to implement mobility management of users in the network.

Unlike other signaling protocols, such as Q.931, ISUP and H.323, the SIP entities communicate between them by means of plain text messages encoded with a syntax similar to HTTP. Text representations of the protocol messages mean additional overhead and processing costs, although they improve readability by humans. It has been shown that call establishment with SIP takes around 1 kB of SIP information. If improved bandwidth efficiency is needed, text compression can be used, but this increases the processing costs even higher. SIP messages are often delivered by using the UDP protocol to avoid TCP connection establishment mechanism. This is not mandatory, and TCP can be used as well.

SIP implements a client/server architecture that has made many IETF protocols, such as HTTP, successful. Every SIP transaction takes the form of a request from a SIP client, and one or more replies from a SIP server. The SIP user agent normally implements both the server and the client. It acts as a client when it is the calling party or as a server when being the called party.

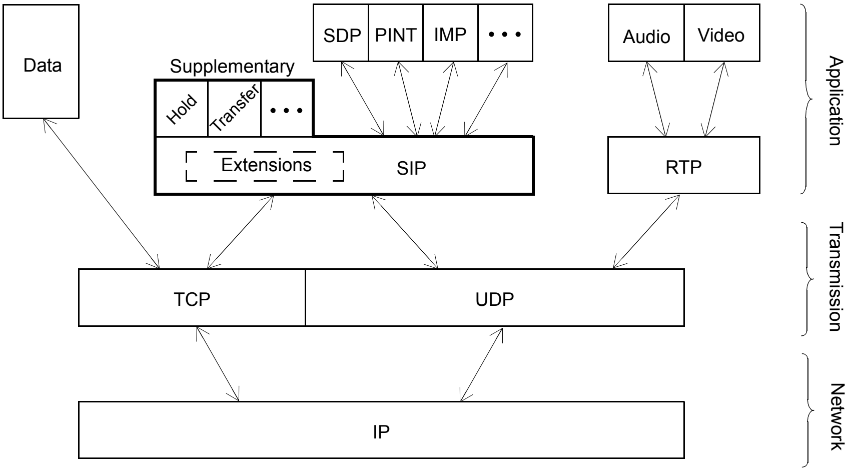


Figure 4.11 IETF multimedia communications protocol architecture

4.3.1 Standardization

The SIP was developed by the IETF *Multiparty Multimedia Session Control Working Group* (MMUSIC WG) and reached the status of a proposed standard in 1999 (RFC 2543). The first standard was replaced by the RFC 3261 in 2002, and it is obsolete today. The version 2 of the SIP protocol was developed by the SIP WG, a group specially devoted to SIP within the IETF.

In addition to the base RFC 3261, there are many complementary RFCs and Internet Drafts for specific applications, supplementary services, interworking and other issues. The standarization effort includes three levels:

1. Maintaining the baseline SIP protocol
2. Developing SIP extensions to support multiple applications

3. Developing features and supplementary services for SIP

As explained, the MMUSIC WG was replaced by the SIP WG. Today, the MMUSIC is in charge of maintaining the *Session Description Protocol* (SDP).

There are other working groups in charge of the IETF architecture for multimedia communications. The most important of these groups are:

- The SIPPING WG identifies and describes the requirements for new SIP extensions. This working group does not implement the extensions, it just identifies and describes them. Then they are delivered to the SIP WG to be included in the SIP RFC or some other standard.
- *SIP for Instant Messaging and Presence Leveraging Extensions* (SIMPLE) WG devoted to development of *Instant Messaging and Presence* (IMP) with SIP.
- *PSTN to Internet Integration* (PINT) WG that addresses the arrangements through which Internet applications can request telephony services.

4.3.2 Architectural Entities

Direct SIP calls between user agents are possible, but often other entities are needed to take advantage of all the features of the SIP. These extra entities make SIP services intelligent allowing user location, supplementary service provision or connection to non-SIP devices. There are other architectural SIP entities as well besides user agents: proxies, registrars, redirect servers and location servers. The purpose of every SIP entity is the following:

- *User Agents* are the devices that initiate requests, being usually their final destination. Some examples are of these are Internet telephones and software phones known as softphones.
- *Proxy Servers* are application-layer routers that forward SIP requests and responses. Their function can be compared to SMTP mail servers.
- *Redirect Servers* provide the location of alternative user agents or proxy servers when the original destination cannot be reached.
- *Registrars* keep track of their assigned network domain. They are in charge of authenticating users, and they maintain information about the subscribers available at every moment.

In practice, it is normal to find proxies, redirect servers and registrars in the same physical device, or even implemented in the same software.

4.3.3 SIP Basic Signalling Mechanisms

The communicating parties interchange SIP requests and responses to establish, end or modify sessions. The transaction model for these messages follows the model of the HTTP protocol. SIP client requests invoke methods with specific purposes in the server. The server processes the request and generates one or several responses with success or fail codes.

As an example, we can think of a user, Alice, that wants to start a VoIP session with a remote user, Bob. Alice calls from a softphone installed on her computer that is connected to the Internet and has a *Fully Qualified Domain Name* (FQDN) *mkt12.flexanet.com.* Bob will receive phone calls in his VoIP phone at *tec26.albedo.com.*

To start the VoIP session, Alice makes use of the INVITE method. INVITE is the most important SIP method. It allows to establish a session with a remote user identified by the user's URI. An INVITE message from Alice with URI *alice@flexanet.com* that wants to call to *bob@albedo.com* could look like this:

```
INVITE sip:bob@albedo.com SIP/2.0
Via: SIP/2.0/UDP mkt12.flexanet.com;branch=z9hG4bK776asdhds
Max-Forwards: 6
To: Bob <sip:bob@albedo.com>
From: Alice <sip:alice@flexanet.com>;tag=1928301774
Call-ID: a84b4c76e66710@mkt12.flexanet.com
CSeq: 314159 INVITE
Contact: <sip:alice@mkt12.flexanet.com>
Content-Type: application/sdp
Content-Length: 142
```

The message starts with the method name, INVITE, the URI of the called party, and the SIP version number.Some header fields follow, with different purposes:

- The *Via* field contains information on the host that is waiting for a response to the INVITE request. It is normally the domain name of this host, and in our example it is the domain name of alice's PC. The Via field also contains information on how to send the response. Our example message states that SIP version 2 must be used. The transport layer protocol must be UDP. The branch number identifies the transaction.
- The *Max-Forwards* field is decremented by one at each SIP-aware hop. It is used to limit the number of hosts the message can traverse in its way to the destination.
- The *To* field contains the URI of the called party, and sometimes an identifying name (Bob).

- The *From* field is an identifying name of the calling party (Alice) with her SIP or SIPS URI. This field also contains a tag parameter that has a random number for identification purposes.
- The *Call-ID* is a globally unique identifier that groups together series of SIP messages that belong to the same signaling dialog.
- The *CSeq* field contains a sequence number (314159) and a method name (INVITE). This serves to sort and identify SIP transactions.
- The *Contact* field contains a SIP URI that represents a direct route to the caller. It must contain an FQDN or an IP address for this purpose. The contact field is useful, because while the first transaction in an SIP session is often routed through a chain of proxies, the transactions that follow can be routed directly between the source and the destination.
- *Content-Type* contains a description of the SIP message body.
- *Content-Length* contains a byte count of the message body.

The INVITE header is not used to negotiate the types of media or codecs to be used. Usually this is done by the SDP protocol that is carried in the SIP message body.

Alice may send the INVITE directly to Bob's phone at *tec26.albedo.com* if she knows where to find him. Otherwise, she could use the location services offered by the SIP proxy of her service provider. In this second case, she sends the INVITE to the proxy at *sip.flexanet.com*. It might be interesting for Alice to route the INVITE via *sip.flexanet.com*, even if she knows the IP address of Bob's phone. Because in this case she can take profit of the ability of SIP to locate users when they change their location. In other words, the SIP proxy can handle mobility. This is not possible with a direct request to the destination.

When *sip.flexanet.com* receives the INVITE from Alice, it tries to find the IP address of Bob's phone or, alternatively, the IP address of another server able to locate Bob. In this example, *sip.flexanet.com* forwards the INVITE to a second sip proxy: *sip.albedo.com.* It adds an additional *Via* field with its own FQDN to the original INVITE request because it wants to receive future notifications about the progress of the call. Then, *sip.flexanet.com* sends a first response back to Alice to inform her that the process of locating Bob has started. SIP responses are similar to HTTP responses. They contain a three-digit code followed by a descriptive sentence. In this case, the code sent to Alice is 100 ("Trying").

When *sip.albedo.com* receives the INVITE, it sends a response with code 100 ("Trying") back to *sip.flexanet.com.* It finds Bob's IP address in a database, and it can then forward Bob the INVITE with an extra *Via* field with its own FQDN.

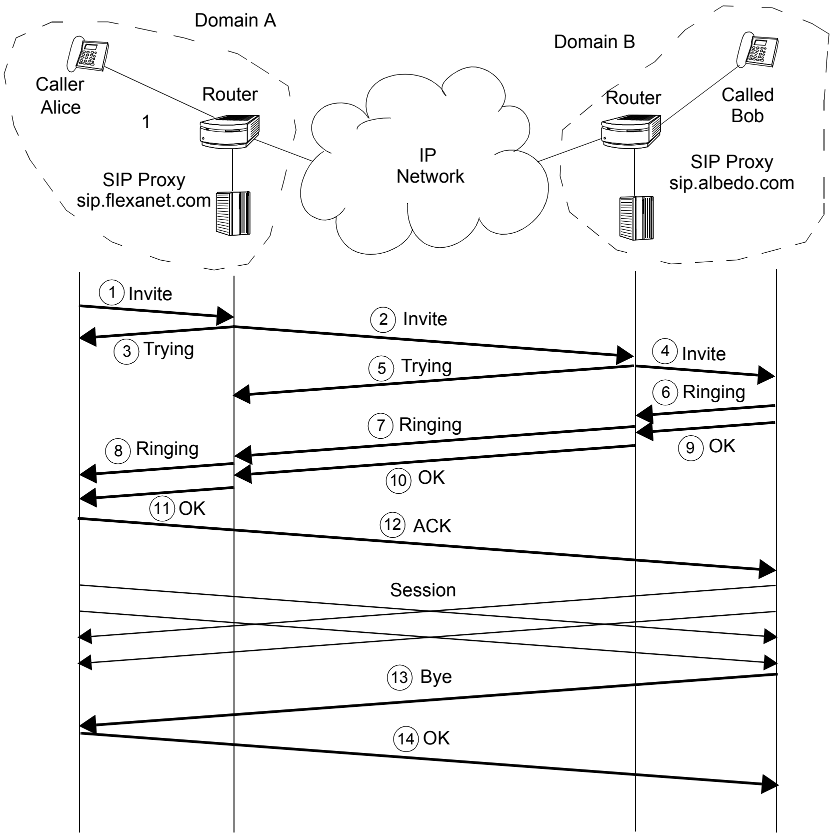


Figure 4.12 Typical SIP session. The calling party does not send the Invite message directly to the called party. It is sent to a SIP proxy that locates the user and starts to negotiate the session parameters.

Bob's phone receives the INVITE from Alice and starts ringing. At the same time, Bob's phone sends back a response with code 180 ("Ringing") to Alice through the chain of SIP proxies, thanks to the extra *Via* fields added by them. If Bob's phone is busy or he decides not to answer, an error response will be sent to Alice. Otherwise Bob's phone will generate a 200 ("OK") response to show he has answered the phone and he is ready to talk. The "OK" response is routed through the chain of proxies, and in this case, it could look as follows:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP sip.albedo.com
      ;branch=z9hG4bKnashds8;received=192.0.2.3
Via: SIP/2.0/UDP sip.flexanet.com
      ;branch=z9hG4bk77ef4c2312983.1;received=192.0.2.2
Via: SIP/2.0/UDP mkt12.flexanet.com
```

```
      ;branch=z9hG4bK776asdhds;received=192.0.2.1
To: Bob <sip:bob@albedo.com>;tag=a6c85cf
From: Alice <sip:alice@flexanet.com>;tag=1928301774
Call-ID: a84b4c76e66710@mkt12.flexanet.com
CSeq: 314159 INVITE
Contact: <sip:bob@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131
```

When Alice receives the "OK" response, she sends an acknowledgement message (ACK) to confirm the reception of code 200 ("OK"). The ACK and all the following transactions in the session are routed directly to Bob's phone, because both Alice and Bob have learned each other's addresses thanks to the *Contact* field of the SIP messages.

After the negotiation phase, the VoIP conversation can start. The SDP handshake usually takes the form of a "two-way" handshake based on a simple request/reply model. After this, the RTP protocol is used to carry the encoded voice samples. The media formats are those exchanged in the SDP handshake.

Both Alice and Bob may decide to modify the session parameters. They can do this by means of a new INVITE message with the new media description. The VoIP session finishes when one of the communicating parties "hangs up" the phone. In this case, a BYE message is generated that is answered by an OK, which ends the session.

4.3.4 The Session Description Protocol

The *Session Description Protocol* (SDP) is an IETF application-layer protocol to describe multimedia sessions. It was specified in 1998, and it is described in RFC 2327. The session description includes:

- Session name and purpose
- Time the session is active
- Media type comprising the session
- Addresses, ports, formats and other useful information to receive and decode the session

The SIP protocol does not provide this service, so it needs the help of the SDP. The SIP is suitable for applications like VoIP. An alternative to SIP for multimedia multiparty session invitation, like video broadcast, is the *Session Announcement Protocol* (SAP). SDP messages can be attached to e-mails and web pages as well. The "application/sdp" *Multipurpose Internet Mail Extension* (MIME) content type allows this kind of attachments. The “application/sdp” is also used by the SIP. Specifically, SDP session descriptions are added to the SIP message body with the “application/sdp” MIME type.

Like many other IETF application layer protocols, the SDP is text-oriented. This is less efficient than a binary encoding, but easier to understand and generate. All SDP messages have a similar format. They consist of a number of lines with the format “item”=”value”. The SDP items are lower case letters of the English alphabet. Some of them are mandatory, others optional. They are always sorted in the same way to recognize transmission errors.

The SDP items are classified in three groups: session description items (see Table 4.3), time description items (see Table 4.4) and media description items (see Table 4.5).

Table 4.3
SDP message items for session description

Order	*Item*	*Description*	*Opt / Mand*
1	v=	Protocol version	Mandatory
2	o=	Owner / creator and session identifier	Mandatory
3	s=	Session name	Mandatory
4	i=	Session information	Optional
5	u=	URL of description	Optional
6	e=	E-mail address	Optional
7	p=	Phone number	Optional
8	c=	Connection information	Optional
9	b=	Bandwidth information	Optional
12	z=	Time zone adjustments	Optional
13	k=	Encryption key	Optional
14	a=	Zero or more session attribute lines	Optional

A session description is made up of one section containing a few session description items, followed by one section with time description items and one or more sections with media description items. A session description might look like:

```
v=0
o=mhandley 28909844526 2890842807 IN IP4 126.16.64.4
s=SDP Seminar
i=A Seminar on the session description protocol
u=http://www.cs.ucl.ac.uk/staff/M.Handley/sdp.03.ps
e=mjh@isi.edu (Mark Handley)
c=IN IP4 224.2.17.12/127
t=2873397496 2873404696
m=audio 49170 RTP/AVP 0
```

```
m=video 51372 RTP/AVP 31
m=application 32416 udp wb
a=orient:portrait
```

In the example, the “v=” item shows the SDP protocol version. The “o=” is a description of the session owner. This is a mandatory field and contains the originating user’s login name (“mhandley”), a session identifier (“28909844526”) that makes the contents of the “o=” item globally unique, a version identifier (“2890842807”) that can be an NTP timestamp to recognize the most recent announcement, the network type, usually “IN” for “Internet”, an address type identifier (IP4) that can be IP4 or IP6 for IP networks, and the originator’s address (“126.16.64.4”).

Table 4.4
SDP message items for time description

Order	*Item*	*Description*	*Opt / Mand*
10	t=	Time the session is active	Mandatory
11	o=	Zero or more repeat times	Optional

The “s=” and “i=” items contain the session name and description given by the originator. The “u=” is an URL that points to a resource with additional information about the session. The “e=” is the E-mail address of the session responsible that does not need to be the session owner. The “c=” item contains connection data. It may contain a unicast or a multicast IP address that is available for transmitting and receiving information in the data session. For multicast sessions, a *Time To Live* (TTL) is appended to the multicast IP address (127 in the example). The “t=” item includes the start and stop times for the session. The values included in this item are decimal representations of NTP time values in seconds.

Table 4.5
SDP message items for media description

Order	*Item*	*Description*	*Opt / Mand*
15	m=	Media name and transport address	Mandatory
16	o=	Media title	Optional
17	s=	Connection information	Optional
18	i=	Bandwidth information	Optional
19	u=	Encryption key	Optional
20	e=	Zero or more media attribute lines	Optional

In our example, there are three media description fields that specify three media streams: one for audio, one for video and a third one is of “application” type. All three have the same format. First, the media type is shown. The options available are "audio", "video", "application", "data" and "control". These are followed by the transport layer port to which the media stream will be sent. This is followed by the transport protocol for the stream. Two options are currently available for this: "udp" if the UDP is to be used, and "RTP/AVP" to specify the RTP protocol. Finally, the media type (as specified in the RTP profile) is included. In our example, the "0" included for the audio stream means that PCM encoding with μ-law scaling is being used, and the 31 for the video stream stands for the ISO/IEC 11172-3 and 13818-3 MPEG-1 / MPEG-2 payload (see Table 4.1). For non-RTP sessions this is not possible. UTP payloads are specified by means of a registered MIME type. In the example, "wb" is used to specify a payload with MIME type "application/wb", the correct one for a whiteboard application.

The defined item set for SDP must not be extended. The "a=" item provides a mechanism to extend and customize the protocol's capabilities. In the example, the "orient:portrait" is used to define the whiteboard orientation (portrait or landscape) in a webminar. Another interesting application of the "a=" is the mapping of dynamic RTP payload numbers to customized payload formats. For example, "a=rtpmap:80 L16/16000/2" would map the RTP payload type 97 to a linear encoded stereo audio stream sampled at 16 kHz.

4.3.5 Security Issues

The SIP header reveals information that the communicating parties may want to keep private. The same can be said about SIP message bodies. This information might be used by a malicious user to access confidential information, change the contents of the session, or fake the identity of someone else. The contents of a SIP message can be secured to avoid these situations. There are mechanisms to ensure SIP message confidentiality, authentication and integrity. Some of them have been specially defined for SIP, while others are of general application in IP networks. These security mechanisms must be combined properly to obtain a trusted service.

The security mechanisms can be classified into end-to-end and hop-by-hop mechanisms. The SIP provides end-to-end protection, but hop-by-hop protection relies on mechanisms external to the SIP, such as IPSec and the *Transport Layer Security* (TLS).

The main advantage of IPSec and TLS is that they make it possible to protect the full SIP message header and body, because they work at protocol layers below the application layer. The drawback of IPSec is that it usually needs special configuration of network nodes. This usually limits the application of this mechanism to servers of the same service provider. TLS is more flexible, but it only works with TCP and is incompatible with UDP.

It must be noticed that if a user agent uses IPSec or TLS to communicate with a proxy, this does not guarantee that the secure transport will be used on the rest of the end-to-end path. The SIPS URIs provide a way to request the use of TLS along the complete path.

4.3.5.1 Authentication

SIP authentication allows a user agent to prove its identity to another user agent, a proxy server or a registrar. Proxy-to-proxy authentication is not provided, and it must rely on IPSec or TLS.

The SIP authentication is a challenge-based mechanism derived from the HTTP Digest authentication. In challenge-based authentication mechanisms, when a server receives a request, it sends a challenge to the client to prove its identity. The challenge is usually a nonce value uniquely generated and used for one challenge only. The client generates a response value calculated from the nonce, and a secret password shared with the server. The server compares the response with an internally calculated value, and the client is authenticated if the response matches this value.

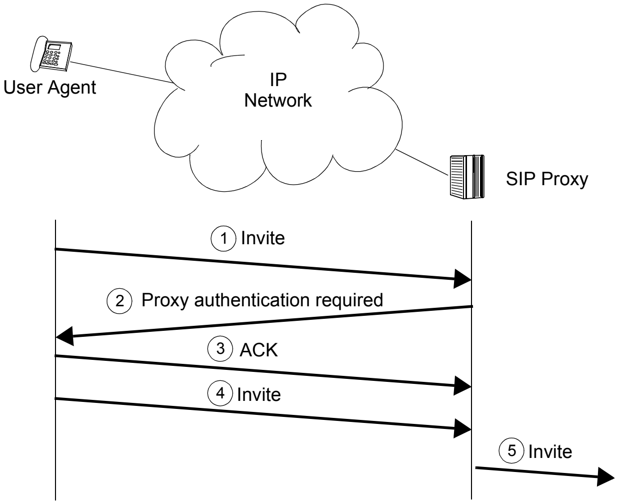


Figure 4.13 Proxy authentication procedure for a user agent

In challenge-based authentication, the password is never sent as normal text. To be secure, it must be difficult to derive the password from the response generated by the client. So, it is encrypted by using some cryptographic hashing algorithms. The default for SIP authentication is the MD5 algorithm.

Some proxies may require authentication. When a proxy server receives a SIP request, it decides whether client authentication is needed or not. If it is, it sends the client a response with code 407 ("Proxy Authentication Required"). The error response is at the same time a challenge, and it contains the nonce and a realm field with information about the registration domain (see Figure 4.13).

Another context where authentication may be necessary is when user agents are registered in SIP registrars. The SIP method needed to do this is REGISTER. Some user agents may require authentication as well. The authentication procedure is similar. The difference is that in this case the error code generated by the remote user agent or the registrar is 401 ("Unauthorized").

4.3.5.2 Confidentiality and Integrity

The end-to-end mechanism to ensure confidentiality and integrity in SIP message bodies is based on the E-mail application. SIP messages carry MIME bodies, so S/MIME can be used to protect them.

S/MIME offers a public-key security framework based on certificates. There are two different types of keys in S/MIME: public and private keys. Every private key has a public key associated with it, but it is very difficult to derive the private key from its public equivalent.

Private keys are used to decrypt and sign messages and must be kept in secret. Public keys are used to encrypt messages and check signatures. The tasks performed by public keys are not sensible, and therefore there is no security risk in distributing them.

S/MIME is vulnerable to personality theft of the legitimate owner of the public key. This is the reason why public keys are always distributed using certificates. Certificates contain public keys, and they are also a way to ensure that the key belongs to its legitimate owner, because they are cryptographically signed by a trusted entity known by all the communicating parties. This guarantees that the key belongs to the entity who issued the certificate. This trusted entity is known as the certification authority.

To be able to encrypt or check the integrity of the S/MIME body of a SIP message, one must request a certificate containing the public key of the transmitter.

The main problem with S/MIME is that it only protects the body of the message – the header remains unprotected. There is a solution for this: An SIP message can be tunneled into an outer SIP message. The original SIP message is sent as an S/MIME body of the outer message. Some of the header fields of the inner message must be copied to the outer message, but the most sensitive data can be hidden within the inner message. The copied fields also provide a way to check the integrity of the outer header fields by comparing them with the inner fields after decryption.

Another inconvenience of S/MIME is that although it is not very common, some intermediate nodes may change the contents of SIP message bodies. If the contents of the body are hidden by S/MIME encryption, these network nodes will not be able to do their job.

4.3.6 Service Architecture and Protocol Extensions

The basic services provided by SIP core methods and headers include locating resources or communicating parties, as well as sending invitations to sessions. This is complemented by the negotiation of session parameters offered by the SDP protocol. These basic services can be extended to offer a broader set of features.

Some of the additional features of SIP are bundled with the basic services, enhancing and adding intelligence to them. These are called supplementary services, and most of them are directly related to the features provided by circuit-switched telephony, including call transfer, call diversion, call hold and call intrusion. There are some other features that are new applications, not provided by the basic SIP protocol, for example SIP *Instant Messaging* (IM).

Supplementary services have not been standarized in SIP, as has been done with the H.323 architecture. Specification lists for new services are written by the SIPPING working group. These specifications are analyzed by other IETF working groups that evaluate whether it is possible to implement the new service with the baseline protocol, or if a specific protocol extension is needed. Protocol extensions are implemented in SIP by defining new methods, new headers and new response codes. The following sections describe some important SIP extensions.

4.3.6.1 Instant Messaging and Presence

Instant Messaging (IM) is exchange of information, usually by means of short text messages, between two or more parties in real time or almost real time. The real time assumption is important to differentiate IM and E-mail; E-mail messages can be delayed by intermediate servers during a long period of time.

IM also includes presence information and “buddy lists” that allow users to get information on the connection status of other users (online, offline, busy and so on). The combination of IM and presence information is very powerful, and sometimes it is referred to as *Instant Messaging and Presence* (IMP).

There are SIP extensions that enable IMP applications based on this protocol, for example the MESSAGE method. For presence, there are two methods that form the event communication architecture for SIP, and they are SUBSCRIBE and NOTIFY.

A SIP request with the MESSAGE method would look like this:

```
MESSAGE sip:bob@albedo.com SIP/2.0
Via: SIP/2.0/TCP mkt12.flexanet.com;branch=z9hG4bK123dsghds
Max-Forwards: 69
To: Bob <sip:bob@albedo.com>
From: Alice <sip:alice@albedo.biz>;tag=49583
Call-ID: asd88asd77a@mkt12.flexanet.com
CSeq: 1 MESSAGE
Content-Type: text/plain
Content-Length: 30

Hi Bob, this is an IM example.
```

The header of the MESSAGE request has fields similar to other SIP requests, such as INVITE. The message body can be of any MIME type, including “message/cpim”, thus enabling interoperatibility of all IMP software supporting this format.

SIP MESSAGE transactions follow the same path as other SIP signaling transactions. This is important, because in the past, IMP applications have registered a high volume of usage, and there is a danger for the SIP-based IMP applications to interfere with other session signaling. To minimize congestion risk, the size of the messages is limited, and the use of transport protocols (like TCP) with some congestion control mechanisms is recommended.

The presence application requires using presence agents that are entities able to store presence information and generate notifications to users when changes occur. Presence agents can obtain presence information from the registrar database, but this is not mandatory. Users interested in receiving presence information about their contacts need to send SUBSCRIBE requests to the presence agent. The presence agent uses the NOTIFY method to inform about andy changes in the “presence” state of the other users.

4.3.6.2 Extensions for Supplementary Services

Supplementary telephony services, like call transfer or call hold, are currently supported by the PSTN, which is why they are an important requirement for IP telephony, especially for SIP.

A simple example of a supplementary service is call hold. Call hold makes it possible for one of the parties to temporarily interrupt (hold) an active call with the other party. There is no need to define anything new to support call transfer with SIP. Call hold can be achieved by a re-invitation: a second INVITE message with a session description set to inactive. The session can be restored after the interruption by sending a third INVITE message with the appropriate SDP description.

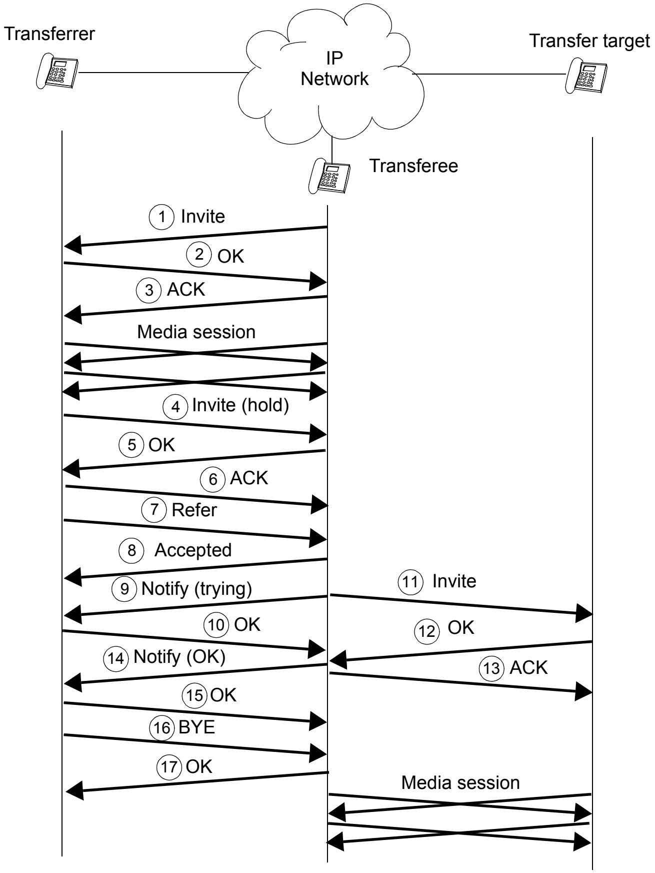


Figure 4.14 Successful call transfer dialog

Call transfer is another supplementary service that makes it possible to transfer an active session from one user to another (see Figure 4.14). This service is implemented in SIP by using the REFER method. A REFER request indicates that the recipient should contact a third party using the contact information provided in the "Refer-To" header field. For example, the following field included in a REFER request means that the recipient should contact *alice@albedo.com*:

```
Refer-To: sip:alice@albedo.com
```

If the REFER request is accepted, the URI identified in the “Refer-To” field is contacted by means of the normal mechanisms provided by SIP, for example INVITE. The NOTIFY method is useful to inform the transferrer on the progress of the transfer. The information is carried in the NOTIFY request body with “message/sipfrag” MIME type. The body of this messages could be like this:

```
SIP/2.0 100 Trying
```

Many variations of call hold and call transfer can be implemented with SIP. Other supplementary services for SIP are call forwarding, 3-way conference, call pickup and automatic redial.

4.3.7 Firewall Traversal

Firewalls isolate trusted networks and hosts from untrusted networks and hosts; in other words they control the traffic between trusted and untrusted hosts. Data packets are allowed to go through depending on the configuration of the firewall.

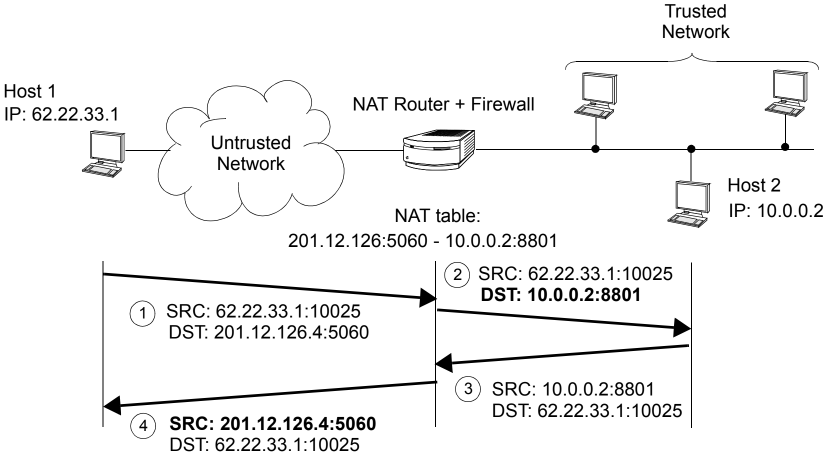


Figure 4.15 A firewall filters traffic between trusted and untrusted networks. NAT further enhances security by hiding private network addresses.

A typical application scenario for a firewall is to protect a private LAN connected to the Internet. In this case, the firewall is placed between the Internet (external network) and the LAN (internal network). Furthermore, the routers that connect private LANs to the Internet often implement some kind of *Network Address Translation* (NAT) that further enhances security and makes it possible to share just a few public IP addresses among many hosts. These hosts are configured with special private IP addresses that are inexpensive and can be managed by the network administrators of the private site.

NAT replaces internal (private) addresses by external (public) addresses. This means that for incoming packets, the router replaces the destination address by a private IP address, and for outgoing packets, the router replaces the source IP address. The router maintains a NAT table that relates external and internal IP addresses (see Figure 4.15).

4.3.7.1 Types of NAT

There are several well-known problems for SIP when crossing firewalls, most of them related with NAT. The complexity of the problem is increased by the fact that there are several types of NATs that exhibit different behaviour with SIP:

- *Full cone NAT*: There is a well-established mapping between internal and external addresses. An external host who wants to send data to an internal host only needs to know the mapping for the address of that host.
- *Restricted cone NAT*: NAT records are temporary, and they are only created when data is sent from an internal address to an external address. The records are removed from the table after being unused for a certain amount of time. External data can pass through the NAT router if the destination IP address matches an external address in the NAT table. For this reason, NAT connections cannot be started by external hosts in the restricted cone.
- *Port-restricted cone NAT*: The port-restricted cone NAT is similar to the restricted cone NAT. NAT records are temporary created when data is sent from an internal address. External data can only pass through the NAT router if both the destination IP address and port match an external address in the NAT table.
- *Symmetric NAT*: NAT records contain not only internal and external addresses, but also destination addresses. Packets with different destinations are translated in a different way, and external computers can only respond to the mappings generated for them.

4.3.7.2 Problems

Problems with NAT arise, because with SIP, there is some addressing information that is carried in the application payload. This information is bypassed by devices that only work at layer 3. Addresses and ports carried by the SIP header and payloads are not translated by the NAT server, so the receivers are not able to find the original sources.

This problem affects both SIP signaling and media transport:

- *SIP signaling problems*: SIP responses may fail to find the way back to the originator of the transaction if the “Via” or “Contact” fields of the SIP requests cannot be resolved to a public IP address. Even if it is possible to find a route to the source, the reply must be directed to the port opened in the NAT router by the request message. This port may be different from the original source port. Sending responses to the original port or any default port should especially be avoided. Finally, response addresses and ports not always remain open “forever” after the first transaction. Most of the NAT routers remove ordinary NAT records from the NAT table after an inactivity period that could range from a few seconds to several hours.

- *Media transport problems*: The media transport protocol, usually RTP, may fail to find the participants of a session if they are behind a NAT router. The reason for this is that the destination addresses for the media session are exchanged by the SDP protocol carried in the payload of SIP packets, and therefore not translated by the NAT router.

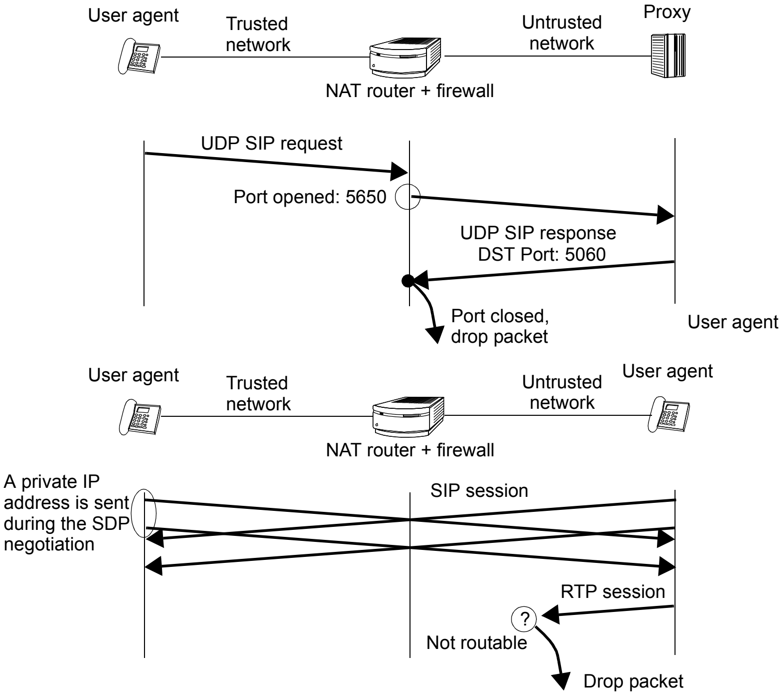


Figure 4.16 NAT translation problems involve both the signaling protocol and the media transport protocol: (a) SIP responses cannot pass through the NAT router, because they are sent to a closed port. (b) RTP packets cannot reach their destination, because the user agent behind the NAT router has sent a private address in the SDP handshake.

4.3.7.3 Solutions for NAT Traversal

There are many possible problematic situations when NAT is used. A typical example would be a user agent that wishes to establish a connection with a second user agent without using a SIP proxy. The first user agent is behind a NAT router. If the receiver sends the response to the IP address included in the "Via" field of the SIP header or to any default port, the communication will surely fail. The best solution is to send the response to the same source address and port received in the request message. These parameters must match the external IP and port of the NAT router.

The problem is more difficult if there is a proxy between the communicating parties. In this case, the session setup cannot continue when the first response arrives to the proxy, because:

1. The source address and the source port of the SIP response are written by the user agent that is in the external network.
2. The connection data of the SIP "Via" field comes from within the private network, and therefore it is not routable in the external network.
3. All other available data ("Contact" SIP field and SDP connection data) is information from the private network, and it is not useful for the proxy.

The SIP protocol has a solution for this problem. There are header tags that help the proxy to find the way to the source:

1. The "received" tag is added by the proxy in the "Via" field when it receives the request from the calling user agent. It shows the IP address of the initiator as it is seen by the proxy. The "received" tag is still present when the proxy receives the response from the called party, and it provides an IP address where the response is sent (as the original SIP-field IP address is not a public one).
2. The "rport" tag works similarly to the "received" tag, but it shows the port to forward responses back to the source.

An INVITE, as it leaves the intermediate SIP proxy, could look like this:

```
INVITE sip:bob@albedo.com SIP/2.0
Via: SIP/2.0/UDP sip.albedo.com
      ;branch=a71b6d57-507c77f2
Via: SIP/2.0/UDP 10.0.0.2
      ;branch=z9hG4bK776asdhds;received=62.22.33.1;rport=5608
To: Bob <sip:bob@albedo.biz>;tag=a6c85cf
From: Alice <sip:alice@albedo.biz>;tag=108bcd14
Call-ID: 4c88fd1e-62bb-4abf-b620-a75659@10.0.0.2
CSeq: 703141 INVITE
```

```
Contact: <sip:alice@10.0.0.2>
Content-Type: application/sdp
Content-Length: 131
```

The “rport” and “received” tags are a good solution for SIP signaling, but the problem with the media transport protocol remains unsolved. For the VoIP application this may mean that the session startup could be successful, but the users may fail to hear the voice from the remote speaker.

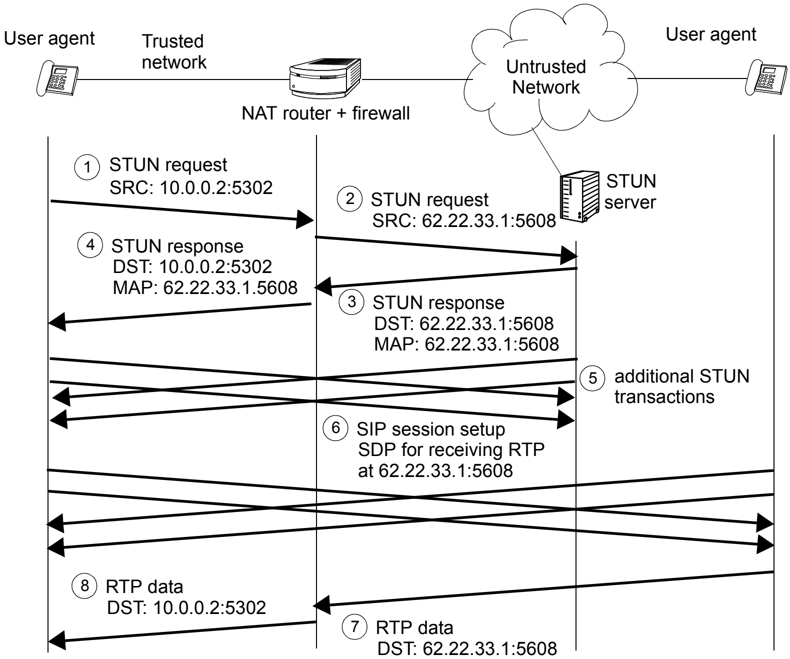

User agent
Trusted network
NAT router + firewall
Untrusted Network
User agent
STUN server
1 STUN request SRC: 10.0.0.2:5302
2 STUN request SRC: 62.22.33.1:5608
3 STUN response DST: 62.22.33.1:5608 MAP: 62.22.33.1:5608
4 STUN response DST: 10.0.0.2:5302 MAP: 62.22.33.1.5608
5 additional STUN transactions
6 SIP session setup SDP for receiving RTP at 62.22.33.1:5608
7 RTP data DST: 62.22.33.1:5608
8 RTP data DST: 10.0.0.2:5302

Figure 4.17 STUN servers help to setup SIP sessions across NAT routers. They give information on the external address and port. This information makes it possible to setup SDP fields with the correct port and IP address to receive media data.

Other type of solutions try to replace the private information of the SIP and SDP fields by public information, meaningful in the external network.

- *Universal Plug and Play (UPnP)*: The UPnP is a solution that allows users behind a NAT firewall to ask the router what will be the NAT mapping for a particular communication. The information received via UPnP is then inserted into the corresponding fields in the SIP and SDP messages. The UPnP mechanism does not work in cascaded NAT connections, and there are security issues that must be addressed.
- *Simple Traversal of UDP through NAT (STUN)*: The STUN is a protocol that allows for external-network queries on how the packets are seen from the out-

side. The information is sent back to the client, and it may be used in the SIP and SDP messages (see Figure 4.17). The main inconvenience of the STUN protocol is that it does not work with symmetric NAT.

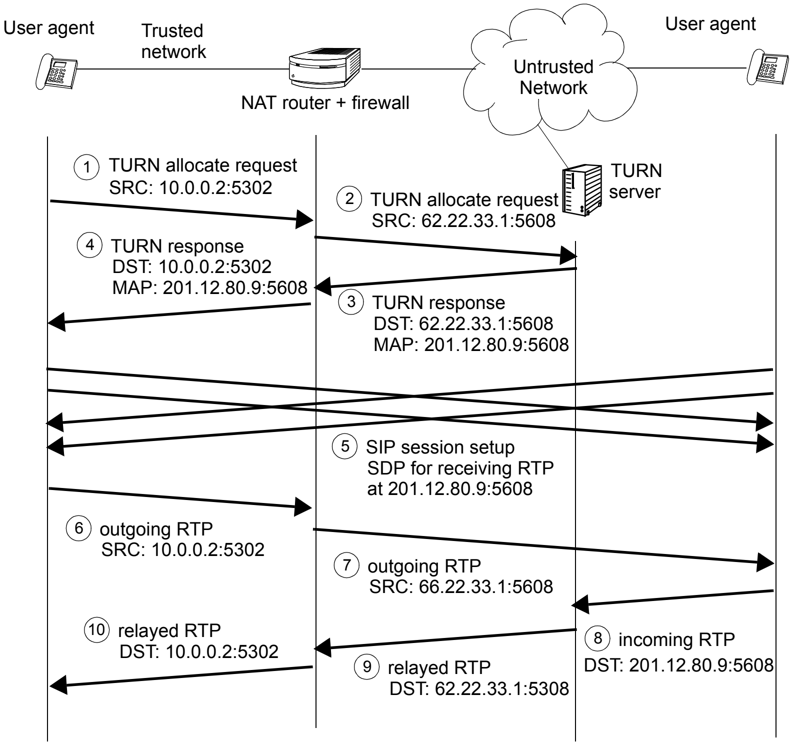


Figure 4.18 TURN servers help to setup SIP sessions across NAT routers. They act as a relay for the incoming media, sent from a remote user agent. They work for all NAT types, but this is not an optimal solution.

- *Traversal Using Relay NAT (TURN)*: Finding a solution for symmetric NAT is more challenging than finding one for other NATs. The STUN protocol is not valid, because the external mapping seen from the STUN server is different from the address seen from the remote SIP entity. TURN is a solution that can be used for all NAT types. A TURN server provides the calling user agent with an IP address and a port that are written in the SDP payload for receiving RTP data. The SIP session setup can then proceed by using the “received” and “rport” tags. After setting the session up, the outgoing RTP data is sent directly to the destination, but the incoming RTP is sent to the TURN server that acts as a relay (see Figure 4.18).
- *Interactive Connectivity Establishment (ICE)*: TURN works with all NAT types, but it must only be used as the last resort, because it wastes resources on

the TURN server. STUN is a good solution, but it is not valid for symmetric NAT. ICE combines both protocols and it gives the best solution for all NAT environments. It gives the client a set of prioritized candidate addresses that can be used for sending and receiving media with the media protocol.

4.3.8 Interworking with the PSTN

Although SIP is a general-purpose protocol for session establishment and tear-down, it has been designed with the VoIP application in mind. It is expected that in the near future, one of the main SIP applications will be VoIP call signaling.

Here, the SIP protocol will have to interact with legacy PSTN signaling and other signaling protocols for VoIP, such as the ITU-T H.323. It is thus necessary to find and implement solutions that make interworking of SIP telephony and other telephony architectures possible. Interworking with PSTN in particular is very important for an smooth transition to a converged voice and data network. For example, it should be possible to establish a call from a softphone to an analog phone or an *Integrated Services Digital Network* (ISDN) phone.

Interworking with PSTN is achieved by using gateways. These gateways are network elements in charge of translating the media between the IP network and the PSTN. PSTN gateways may accept analog or digital telephony interfaces. Some of them may need to work with TDM interfaces, like ISDN *Primary Rate Interfaces* (PRI). The signaling associated with the media needs to be translated as well. While SIP is likely to be widely deployed for telephony applications over IP, the dominant signaling framework for the PSTN will still be based on *Signaling System 7* (SS7).

Strictly speaking, SS7 is not a signaling protocol. It is more a suite of protocols working together. Within the SS7 architecture, the ISUP is a popular alternative to establish and tear down voice calls. PSTN gateways may try to translate ISUP messages into SIP messages to enable signaling interworking between the PSTN and the IP network (see Figure 4.19). For example, a gateway may translate a SIP INVITE request into an ISUP *Initial Address Message* (IAM[1]), or an ISUP *Address Complete Message* (ACM[2]) can be translated into a 183 (Session Progress) response code, and an ISUP *Answer Message* (ANM[3]) might be translated into a 200 (OK) response code.

[1] IAM is the ISUP message that contains the calling number, the called number, the bandwidth requirements for the communications, the type of caller, and other information.

[2] ACM is the ISUP message that indicates that the call is connected to the destination. This message is a response returned when an IAM is received. It causes a one-way audio to be opened from the destination switch so that the caller can hear a ring-back tone.

[3] ANM is the ISUP message sent towards the originator when the called party answers the phone.

Translation between signaling messages is not always straightforward. Sometimes, translated messages are only approximations of the original messages. This makes signaling interworking a complex problem.

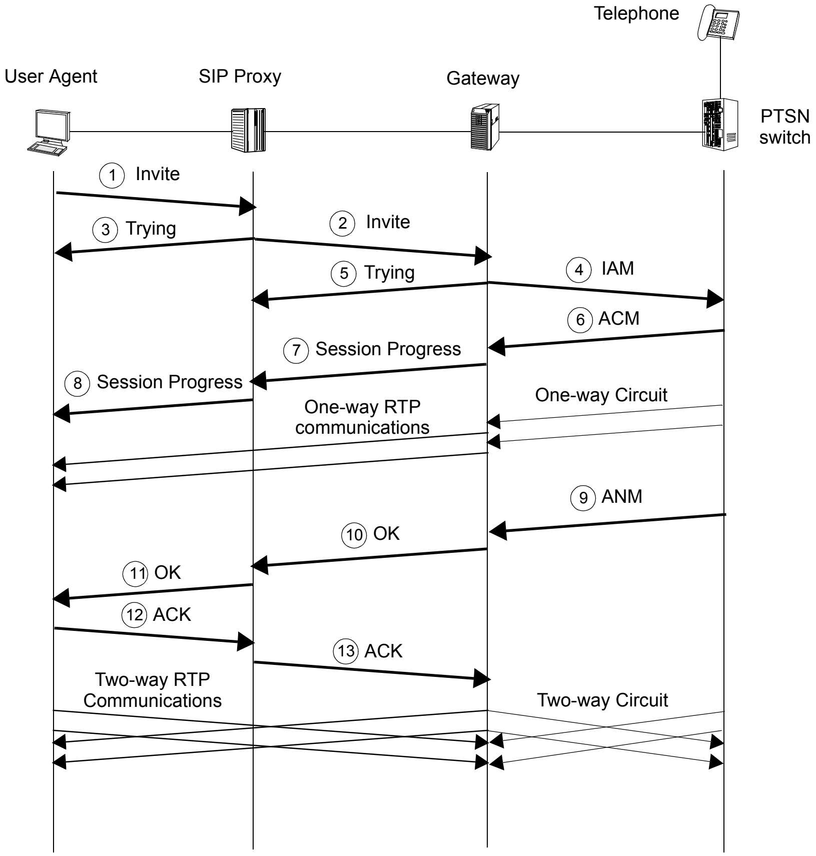


Figure 4.19 The gateway translates signaling messages between the PSTN and the IP signaling protocols

4.3.8.1 Gateway Decomposition

PSTN gateways must be able to perform complex signal processing operations on large volumes of voice traffic.They must also translate PSTN and IP signaling, and provide services. Signal processing tasks can be handled efficiently by digital signal processors, but this hardware is not well suited to implement protocol stacks or run general-purpose code.

This is the reason why it has been proposed to divide the PSTN gateway into two separate network elements (see Figure 4.20):

- *Media Gateways (MG)* handle protocol, format and encapsulation conversions on the media data.
- *Media Gateway Controllers (MGC)* handle all tasks related with call control and signaling translations.

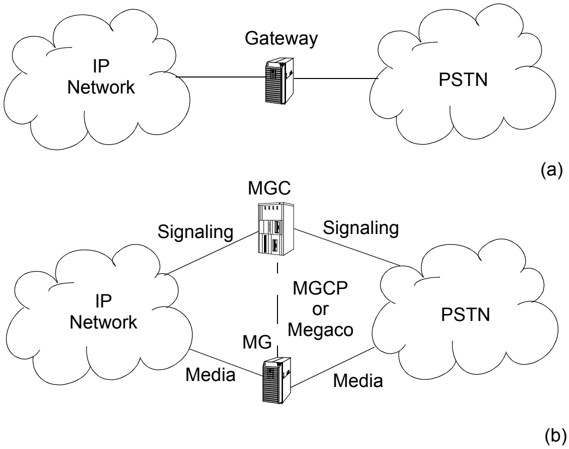


Figure 4.20 Gateway decomposition. The functions of the PSTN gateway are shared by two network elements: (a) a single network element supports all tasks related to interworking; (b) the MG translates between PSTN and IP media formats, and the MGC translates PSTN and IP signaling formats.

The gateway decomposition is a step beyond the separation of media transport and signaling transport specified in the IETF VoIP model. Signaling and media may be delivered by different paths in the network and are processed by specialized network elements. It is even possible to send signaling and media across different networks.

Network intelligence is provided by means of signaling, and thus it is normal to think that the MGCs are intelligent network elements. This results in a network model where MGCs are active network elements and MGs remain mostly passive, controlled by MGCs. A protocol is needed to establish dialogs between MGCs and the MGs controlled by them. The requirements for such a control protocol are:

- *Media session management.* The protocol must enable the MGC to establish, end or modify media sessions in one or several MGs remotely.
- *Transcoding management.* It should be possible for the MGC to specify the transformations to be made to media streams as they pass through an MG.

- *QoS awareness*. It should be possible to specify QoS in media sessions in terms of delay, loss probability or other performance parameters. The MG must be able to report QoS statistics to the MGC for network management purposes.
- *Embedded signaling management*. Signaling information may be embedded in the media stream. An example of this are the *Dual Tone Multi-Frequency* (DTMF) tones. It must be possible for the MG to report the occurrence of this kind of signals. The MG must also be able to request information about embedded signaling, or be configured to perform certain actions when certain signaling events occur.
- *Generating and managing billing information*. It must be possible to report billing and accounting information. This information may include the start and end times of media sessions, or the transmitted volume of information in the media session.
- *Flexible and scalable MG management*. It must be possible to assign associations between MGCs and MCs with flexibility. Certain functions, like back-up MGC designation are needed for a fault-tolerant system. MGCs must be able to deal with different MGs, as there are many different transmission technologies and heterogeneous implementations by different equipment manufacturers.

The IETF has developed the *Media Gateway Control Protocol* (MGCP) to deal with MG / MGC interconnection. This protocol is specified in RFC 2705. There are many commercial solutions based on the MGCP, but the preferred solution for new implementations is the Megaco/H.248 protocol, developed jointly by the IETF and the ITU-T.

4.3.8.2 Converged Telephony

VoIP has revolutionized the world of telephony. It is currently possible to perform voice calls across the Internet at no cost whatsoever. Internet VoIP calls do not meet the minimum requirements of a carrier-class service (high availability, quality and connectivity), but new voice operators are entering the scene anyway, offering VoIP services at very low cost. This new scenario is pushing the traditional telephone operators to adopt IP solutions to transport voice.

MGs and MGCs are critical for telephone operators to migrate their voice networks to IP while maintaining full connectivity between their subscribers. There are two types of MGs that can be used:

- *Trunking MGs*. These are MGs that connect the PSTN with the IP network. They have to deal with SS7 or other type of *Network-to-Network Interface* (NNI) signaling and manage a large number of circuits.

- *Access MGs*. These MGs connect User-to-Network Interfaces to the IP transit network. They may have analog or ISDN interfaces (PRIs or BRIs) to the user installations.

Access MGs enable telephone operators to migrate to a packet-based transit telephone network without replacing the analog or ISDN subscriber telephones by VoIP telephones (see Figure 4.21).

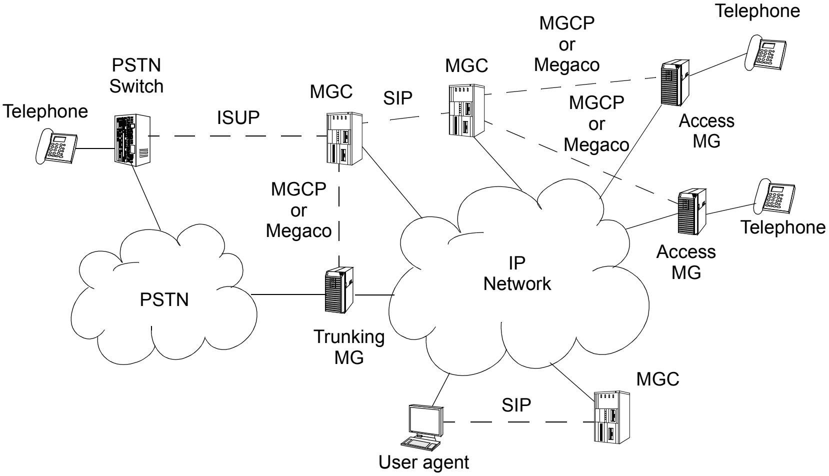


Figure 4.21 Converged VoIP architecture: Telephones are connected to an IP network, thanks to residential MGs that are able to translate voice in a format that can be transmitted over a packet network. PSTN and IP networks can be connected by larger trunking MGs. All these gateways are controlled by MGCs. Communications between MGs and MGCs use MGCP or Megaco. SIP is used for communication between different MGCs.

In a network architecture with MGs, native VoIP user agents can establish voice calls with analog or ISDN phones. To do this, the VoIP user agent (directly, or with the help of one or several SIP proxies) establishes a SIP session with an MGC. The MGC then forwards the media through an access or trunking MG. The media may have to travel trough the PSTN. To make this possible, an MGC must translate the SIP signaling into a PSTN signaling language, such as ISUP. In this network architecture, MCGs are seen as normal SIP user agents that establish, modify or end PSTN telephone sessions on demand. Like any other SIP network element, MGCs deliver information between them by using the SIP protocol.

Due to the convergence between PSTN and IP networks, two PSTN telephones could be exchanging information between them without noticing that the voice is actually being delivered via an IP network.

Chapter 5

IP Multicasting

In the 1990's, the most important Internet application was the *World Wide Web* (WWW), a unicast client/server application. For the WWW, an application-layer protocol called *Hypertext Transfer Protocol* (HTTP) is used by the client to request information from a specialized server. This server then sends HTTP answers to the client. Another popular application for the Internet is e-mail, where one sender delivers mail messages to the mailboxes of one or more recipients using an application-layer protocol, the *Simple Mail Transfer Protocol* (SMTP). The e-mail architecture is quite different from the WWW. E-mail is also a client/server application, but it uses multicasting. In this case, the problem of multicast delivery is solved by SMTP. From the point of view of the network, the paths to the different recipients of an e-mail are seen as different routes by the intermediate routers.

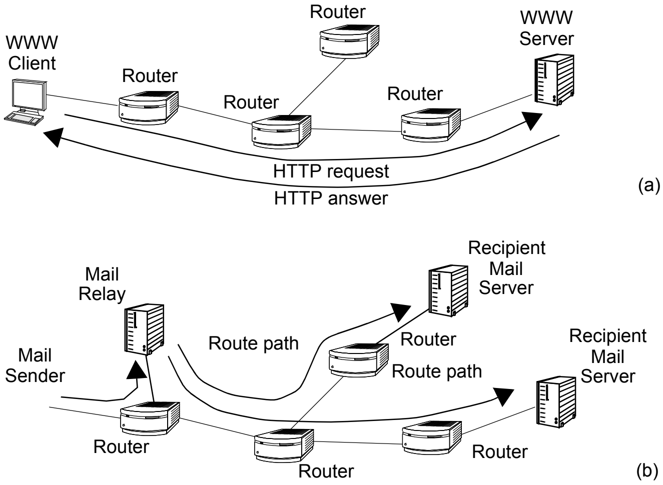


Figure 5.1 Service model for IP data applications: (a) The WWW service model: Unicast and client/server-based. (b) The e-mail service model: Multicast application; multicast delivery is solved at the application layer.

Today, the convergence of circuit-switched and packet-based technologies is making it possible to deliver voice, video and data using one single IP network. Many of the new multimedia applications, such as IPTV or VoIP are multicast, but the multicast service model of the e-mail application is not enough for these applications. It does not have the scalability and simplicity needed to deliver real-time voice or video to hundreds or maybe thousands of receivers. To make an IP network support triple play, it is necessary to implement multicasting in the network layer of the protocol stack. This means that multicast delivery must be supported by intermediate routers as well, not only by terminal equipment.

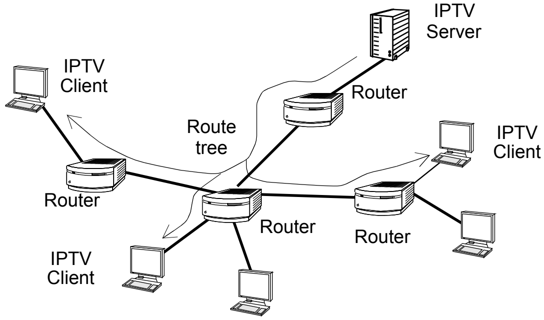


Figure 5.2 Service model for the IPTV application. Multicast delivery is solved at the IP layer and not at the application layer of the protocol stack.

The delivery of voice and video makes it necessary to install multicast routing protocols in IP routers. The aim of these protocols is to find routing trees to send multicast data to the destination; in other words, choose the correct outgoing interface for incoming multicast packets. The second problem to be solved is how to manage multicast groups. This means implementing a mechanism to add and remove recipients dynamically. This is done by using multicast addresses and the *Internet Group Management Protocol* (IGMP).

5.1 IP Multicast Groups and Their Management

In the Internet and in any IP network, multicasting means the transmission of IP datagrams to groups of hosts called multicast groups. Every multicast group is identified by a single IP address. Most of the IP addresses are used to identify hosts in the network, but some of them are reserved not for hosts but for groups of hosts. The multicast IP addresses are Class D addresses, which means that they all start with "1110". Expressed as decimals, these addresses fall in the range from 224.0.0.0. to 239.255.255.255.

Table 5.1
Internet address classes

Address class	*First byte (binary)*	*Address range (decimal)*	*Number*
Class A	0xxxxxxx	0.0.0.0 ~ 127.255.255.255	2,147,483,648
Class B	10xxxxxx	128.0.0.0 ~ 191.255.555.555	1,073,741,824
Class C	110xxxxx	192.0.0.0 ~ 223.255.255.255	536,870,912
Class D	1110xxxx	224.0.0.0 ~ 239.255.255.255	268,435,456
Class E	1111xxxx	240.0.0.0 ~ 255.255.255.255	268,435,456

The *Internet Assigned Numbers Authority* (IANA) is in charge of the administration of the Internet addressing space, including multicast addresses. Some addresses within the Class D range are reserved by the IANA for specific applications, and others are dynamically assigned to transient multicast groups.

Those hosts that need to send data to a multicast group simply put the right multicast address in the destination address field of the IP datagrams sent to this group. There are not many requirements for the transmitter, but IP multicasting must be supported by the network, and the IGMP must be supported by the receiver in order to join or leave multicast groups. Supporting multicasting in IP networks is not mandatory; there are several compliance levels that range from no compliance at all to full compliance.

Table 5.2
Some IP multicast addresses reserved by the IANA

Address	*Receivers attached to the group*
224.0.0.1	All Systems on this Subnet
224.0.0.2	All Routers on this Subnet
224.0.0.5	All OSPF Routers
224.0.0.6	Designated OSPF Routers
224.0.0.9	RIP2 Routers
224.0.0.12	DHCP Server / Relay Agent
224.0.1.1	Network Time Protocol

There is no obligation for the members of a multicast group to be in the same network. However, in a very simple IP multicasting model, all the members of the group are in the same Ethernet network with the sender. In this case, the IGMP is not needed, and it is possible to take advantage of the broadcast nature of Ethernet.

If not all the members of the multicast group are in the same network, a *multicast agent* is needed. This is normally a router that delivers the data to the correct destination. The routers in the multicast IP network must use a multicast routing protocol to be able to perform this task.

5.1.1 Multicasting in Ethernet Networks

The IANA controls the block of Ethernet MAC addresses between 01-00-5e-00-00-00 and 01-00-5e-7f-ff-ff. When transmitting IP multicast data in an Ethernet network, the destination address can be automatically mapped into a multicast Ethernet address. The 28 least significant bits of the IP address are copied to the IANA-managed Ethernet address block explained above. For example, the multicast IP address 239.255.0.1 would be mapped into 01-00-5e-7f-00-01 for Ethernet.

The information arrives to all the hosts in the network, and those belonging to the correct group can recognize the multicast Ethernet address and process the correct frames. Frames arriving to those hosts who do not belong to the group are discarded.

Note that in the mapping process, five bits of the IP address are lost. This means that there are 32 IP multicast addresses with the same Ethernet address. This does not have to be a problem, if the use of IP multicast addresses is planned carefully.

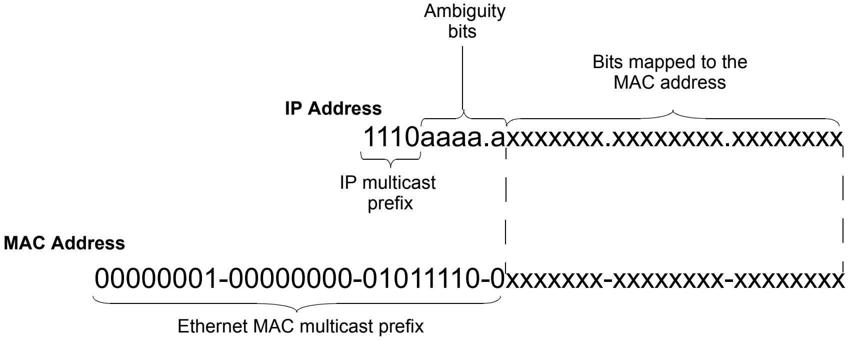

Ambiguity bits
Bits mapped to the MAC address
IP Address
1110aaaa.axxxxxxx.xxxxxxxx.xxxxxxxx
IP multicast prefix
MAC Address
00000001-00000000-01011110-0xxxxxxx-xxxxxxxx-xxxxxxxx
Ethernet MAC multicast prefix

Figure 5.3 Mapping of IP multicast addresses into Ethernet multicast MAC addresses

Ethernet multicasting works in a similar way in broadcast Ethernet networks with shared transmission media and in switched networks with dedicated media. The reason for this is that multicast traffic is broadcast by switches by default. In other words, switched Ethernet behaves the same way as broadcast Ethernet for multicast traffic. This could be a problem, especially in large networks, because multicast Ethernet traffic is directed to hosts that do not request this traffic. In bridged carrier-class networks this may have security implications as well.

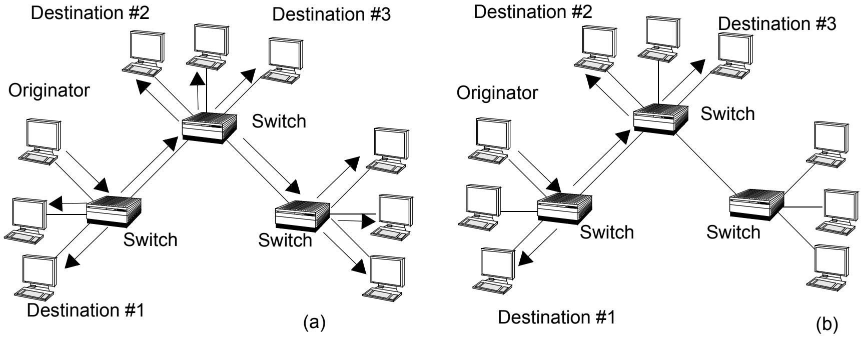


Figure 5.4 (a) Switches not implementing IGMP snooping. (b) Switches implementing IGMP snooping.

It is possible to avoid broadcasting of multicast Ethernet addresses, if IGMP snooping is implemented in the switches. IGMP snooping means that layer-2 switches can check (snoop) layer- 3 IGMP messages. This way, the switches know which are the active multicast groups and who belongs to which group, and they can forward multicast data to the correct ports.

5.1.2 Multicasting and the Internet Group Management Protocol

The IGMP is the network protocol that allows hosts to join or leave multicast groups, and therefore it must be implemented by all hosts that wish to receive multicast information.

An important component of the IP multicasting architecture is the multicast agent. This entity is in charge of:

- *Group management*: The multicast agent grants or denies multicast group membership to hosts. It finds and assigns IP addresses and access keys to the groups and removes inactive hosts from them.
- *Multicast routing*: The multicast agent forwards multicast traffic to remote networks. To do this, it associates IP addresses to sets of interfaces. It is not necessary to maintain a full list of members (and their IP addresses) for every group; routing can be performed just by knowing the network interfaces where the information needs to be forwarded to.

IGMP messages are transmitted between hosts and multicast agents that establish a client/server relationship. Hosts are clients and multicast agents are servers. IGMP packets have the same format, defined by the IETF to include the following fields:

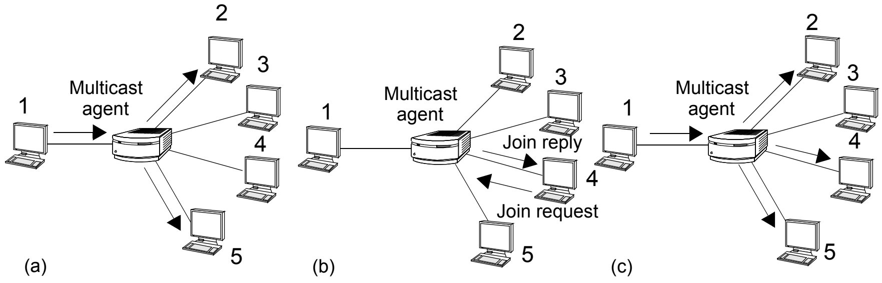


Figure 5.5 Joining a multicast group with IGMP: (a) The multicast agent routes traffic only to the members of a multicast group (hosts 2 and 5). (b) Host 4 requests joining the multicast group, and the multicast agent grants membership. (c) Now, the multicast agent routes the traffic to hosts 2, 3 and 5.

- *Type*: Describes the purpose of the IGMP packet. IGMP messages can be used to join or leave groups, create new multicast groups, or confirm membership to a particular group.

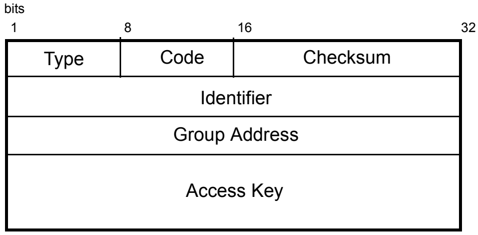
- *Code*: This field is only meaningful when creating a new group. It shows whether the group is going to be public (value 0) or private (value 1). Public groups can be joined freely, but private groups are protected with an access key. In IGMP replies sent by multicast agents, the Code field shows whether the request is granted, denied or pending.

- *Checksum*: This field is for error detection purposes in IGMP messages. It is the one's complement of the one's complement sum of the IGMP message. The Checksum field itself is considered to be zero for this calculation.

- *Identifier*: It is useful to distinguish between the different request messages arriving from the same host. This is why different IGMP messages from the same host have a different Identifier field. When the multicast agent sends a reply, it uses the same identifier as the request message.

- *Group Address*: The group address is assigned by the multicast agent; thus, when creating a new group, the address is sent to the host in the Create Group Reply message. When managing group membership, the IP address of the group to be joined or left is sent in the join/leave request message.

- *Access key*: This field is only meaningful when managing private groups. The access key for private groups is assigned by the multicast agent in the Create Group Reply message. The host needs to supply this key in the request message in order to join or leave a private group.



Type	Message
0x01	Create Group Request
0x02	Create Group Reply
0x03	Join Group Request
0x04	Join Group Reply
0x05	Leave Group Request
0x06	Leave Group Reply
0x07	Confirm Group Request
0x08	Confirm Group Reply

Figure 5.6 IGMP message format

5.2 Multicast Routing

Implementing multicast in broadcast networks such as Ethernet is not problematic, but defining and deploying multicast mechanisms for switched networks such as IP is rather challenging.

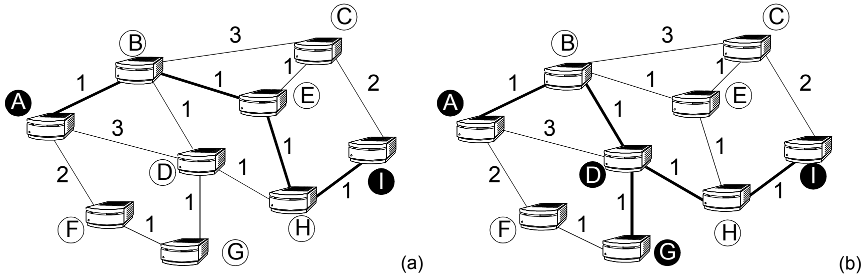


Figure 5.7 Unicast and multicast routing. The "weight" of the links is identified by numbers: (a) The aim of unicast routing is to find routes between a single source and a single destination across a network. (b) Multicast routing tries to find routes to enable communication between multiple transmitters and receivers.

IP multicasting across the Internet was first demonstrated in 1992 by using the Multicast Backbone (MBone). The MBone is an overlay network built on the top of the Internet technology. Several "islands" supporting multicast routing are connected between them by point-to-point tunnels configured in the unicast Internet to allow long-distance multicast communications.

The objective of unicast IP routing is to find routes between a single origin and a single destination in a network. Usually, these routes are paths across several routers. Not all the possible paths between a source and a destination have the same performance. Some paths have less delay than others, and some paths may be more reliable or cheaper. Routes can be manually configured by the network administrator, but for large networks with many routers, it is better to let one or more routing protocols do the job automatically. These protocols make use of routing algorithms to find the best routes. To do this, they need to consider one or more of the performance parameters mentioned before (see Chapter 6 *"QoS in Packet Networks"*). Routing algorithms usually define a weight function that uses numbers to represent the convenience of each link (the better the link, the less weight it has). Then they use an optimization process to find the shortest routing paths (those with minimum weight).

Multicast routing has the same problems as unicast routing. Multicast routing protocols make use of special multicast routing algorithms to find the best route. The difference is that the route is not normally a path, because there are always two or more recipients. Multicast routes are defined as trees where the information flows from one or more origins to several destinations. Multicast routing algorithms are in charge of finding the best possible trees (see Figure 5.7).

Routing requirements differ depending on the multicast service to be transported. Multicast services can be classified in source-specific and group-shared services:

- *Source-specific service*: There is only one transmitter and two or more receivers. An example of a source-specific multicast application is IPTV.
- *Group-shared service*: There are several transmitters that can also be receivers. A good example of a service of this type is a multiparty video conference.

Routing trees can also be classified into point-to-multipoint and multipoint-to-multipoint trees. Source-specific trees have a well-defined root node, but group-shared nodes do not have this property. Point-to-multipoint trees are better suited to source-specific services, and multipoint-to-multipoint trees are used for group-shared services.

5.2.1 Multicast Routing Algorithms

Routing protocols try to find the optimal multicast tree. Unfortunately, the word “optimal” is not a very clear concept. The convenience of some routes over others can be evaluated by means of some of the properties of the trees:

- *Delay*: The delay of a point-to-point path is the sum of the delay of the links that make up the path. Defining the delay for multicast trees is a little bit more complex. For point-to-multipoint trees, delay can be defined as the average path delay for every possible destination. For multipoint-to-multipoint trees, the delay is the average of all the possible point-to-multipoint delays when every end-router is considered to be the root of a tree. For example, if in

Figure 5.7(b) we consider that the label written at each link is the delay for that link and it is accepted that Router *A* is the root router of the multicast tree, the point-to-multipoint delay for this tree is 3. If the same tree is considered to be a multipoint-to-multipoint tree, the delay is 2.55.

- *Scalability*: The network should support many simultaneous multicast trees, and this places some requirements to the trees. The load of a router in the tree depends on the number of copies to make for every incoming packet to deliver data to the different destinations of the tree. This is the reason why trees that tend to distribute load fairly between routers are better than those that tend to concentrate load on a few routers. Of course, scalability is not only a requirement for multicast trees, but also for the routing protocol. The routing algorithm should be able to compute multicast trees in a reasonable amount of time, and the signalling traffic should remain as small as possible. These properties should scale well with the number of nodes in the network.
- *Support for dynamic multicast groups*: Good multicast groups should allow group members to join or leave the multicast tree in a seamless way, without changing the properties of the tree.
- *Fairness*: A multicast tree should not "punish" one multicast group member in order to improve the service for other members. For example, in a multiparty teleconference, it is important that all the receivers hear the speaker at the same time, or in a network game, if the delay is not the same for all the players, the game is no longer fair. This means that fair distribution of delay for different receivers is a condition for multicast trees.

These and other properties can be taken into account by multicast routing algorithms. There are many algorithms, and each of them has its own way to calculate routes. Many of the principles for unicast routing algorithms are valid for multicasting as well. It is a common practice to assign a positive number for each link (known as the "weight" of the link) and then try to find the trees that "weigh" less. These trees are called *Steiner trees*, and they are the same as the shortest paths for unicast routing algorithms. Specifically, a Steiner tree is a tree that:

1. Spans all the all the nodes to be interconnected.
2. Has the minimum weight. The weight of the tree is the sum of the weights of the links.

The tree shown in Figure 5.7(b) is an example of a Steiner tree that spans nodes *A*, *D*, *G* and *I*. Steiner trees can be used as multipoint-to-multipoint trees to enable group-shared services.

Mathematically, it can be shown that there are no good[1] algorithms for finding Steiner trees. This means that none of the possible algorithms for finding Steiner trees scale well for networks with a high number of nodes, and efficient ways for finding steiner trees can only be found for some network topologies, as well as for the following cases:

- *Unicast*: There is only one transmitter and one receiver. The problem is how to find the shortest path between two nodes, and it can be solved with the help of the Dijkstra and Bellman-Ford algorithms.
- *Broadcast*: All the nodes in the network belong to the tree. The problem is how to find the minimum-weight spanning tree for the network, and this problem can be solved by means of the Prim and Kruskal algorithms.

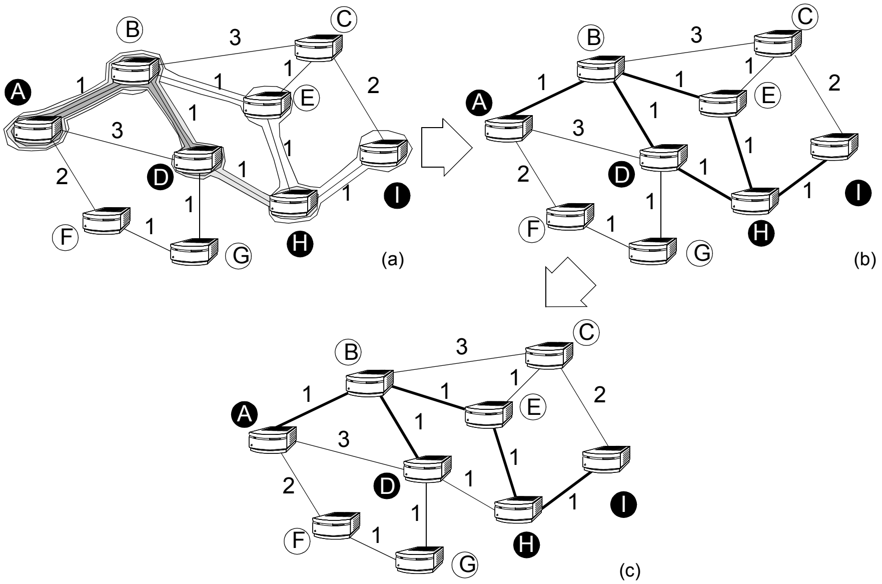


Figure 5.8 Algorithm for finding the minimum-delay point-to-multipoint tree: (a) All the shortest paths are calculated. (b) The union of all the shortest paths is calculated. (c) All the loops are removed to obtain a tree.

Although there are no efficient algorithms for finding Steiner trees, there are some practical approaches that produce good-quality (although not always optimal) trees. Specifically, there are algorithms to find trees with weight that is at worst twice the weight of the optimal Steiner tree. One of these algorithms was proposed by Kou, Markowsky and Berman and is know as the KMB algorithm.

[1] The word “good”, here, means that the problem of finding Steiner trees cannot be solved in polynomial time by any algorithm.

Some other algorithms do not try to find trees similar to the Steiner trees, but take a different approach. Some try to find minimum-delay trees. Algorithms for finding minimum-delay multipoint-to-multipoint trees have the same problem as algorithms for finding Steiner trees, but there is an easy way to calculate point-to-multipoint minimum-delay trees that is used by many routing algorithms:

1. Calculate the shortest path from the root node to every destination node.
2. Calculate the union of all the previously calculated shortest paths.
3. Remove any loops from the result.

Sometimes the quality of a multicast tree cannot be evaluated by simply calculating its weight. Some of the conditions that a multicast tree must meet may affect the routing algorithms. For example, finding a minimum-delay tree may not be enough; it may be necessary to find a tree with balanced delay between all the destinations (delay-variation-bounded multicast tree), or a tree that distributes the load of every router fairly (degree-bounded multicast tree).

Chapter 6

QoS in Packet Networks

Quality of Service (QoS) is the ability of a network to provide services with predictable performance.

Circuit-switched networks have always offered a good QoS when it comes to delay. Especially the end-to-end delay of these networks is usually small and highly predictable. This makes them very attractive for interactive communication services, such as telephony. The drawback of circuit-switching technologies is that they waste a lot of network transmission capacity, because their resource reservation mechanisms are not very flexible. Packet-switching technologies, however, are very flexible, and they use network infrastructure efficiently, but it is difficult to predict their performance in terms of delay, due to variable queuing and processing times in intermediate nodes. The properties of packet-switched networks have made them a good choice for low-cost data communications.

The trade-off between performance and efficiency in networking technologies arises when a single converged network must support interactive and non-interactive traffic. This is what happens with triple-play applications. An important goal for triple-play providers is to base their services on a single network that can transport voice, video and data, to generate revenues while reducing capital and operational expenses (CAPEX and OPEX). This network is ideally based on a single networking technology, but it can offer the best of circuit- and packet-switching technologies.

6.1 QoS Basics

Today's converged networks are based on the *Internet Protocol* (IP). IP is more than 30 years old, but it has a successful history. Most of the current computer applications, as well as the Internet are based on IP. The original masterminds of IP designed it as an ordinary packet-based technology in many aspects. IP networks are very flexible and efficient, but they do not have the minimum delay guarantees needed for high-quality audio and video services. In other words, legacy IP networks can only offer best-effort services.

It has been necessary to update IP so that it supports QoS, and this evolution still continues. Future solutions must take these two facts into account:

1. A network with small packets experiences shorter delays and is thus more similar to a circuit-based network. The reason for this is that a packet cannot be sent before it is filled with data. And, obviously, the time needed to fill smaller packets is shorter.
2. Under light load conditions, packet-based networks look more similar to circuit-based networks. There are no variable queuing times, and end-to-end delay becomes a deterministic quantity.

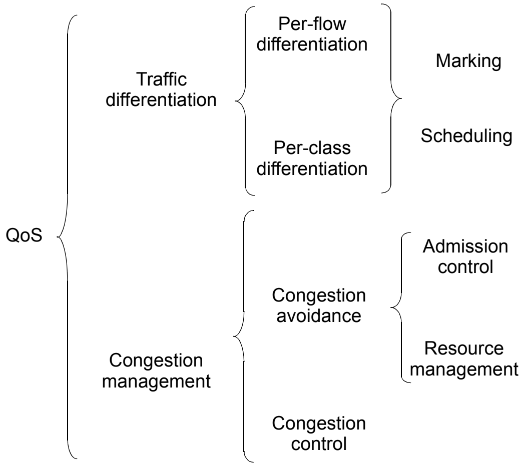


Figure 6.1 It is difficult to achieve good QoS features with one single mechanism. The best way is to mix many elements to get the desired result.

Having small packets in IP networks is not very difficult, because the length of IP packets is variable, and it can be chosen by the transmitter. Another advantage of this variable packet length technology is that packets for data communications can be very long, so that they could achieve a low overhead-to-data ratio, while packets for interactive data and video can be short, to achieve low delay.

Keeping light load in the network is not always easy. Rather than to limit the amount of load, the solution has been to increase the transmission bandwidth to keep network utilization low. This is a good solution when bandwidth is cheap – otherwise it is necessary to find a way to keep delay low and predictable while improving network utilization to the maximum. The current networking technology achieves this by using two features that improve the traditional packet-switching technology:

1. Traffic differentiation

2. Congestion management (control and avoidance)

There are plenty of techniques associated with both traffic differentiation and congestion control and avoidance (see Figure 6.1). The aim, a QoS-aware packet network, can only be accomplished by combining several of these techniques.

6.1.1 Traffic Differentiation

Traffic differentiation separates the bulk traffic load into smaller sets, and treats each set in a customized way. There are two issues related to traffic identification:

1. *Traffic classification*. The traffic is divided into classes or flows. Sometimes it is necessary to explicitly mark the traffic with a *Class-of-Service* (CoS) identifier.
2. *Customized treatment of traffic classes and flows*. Some packets have more privileges than others in network elements. Some may have a higher priority, or there may be resources reserved for their use only.

Traffic differentiation makes it possible to improve performance for certain groups of packets and define new types of services for the packet-switched network.

- *Differentiated services*. We can talk about differentiated services when a part of the traffic is treated ‘better’ than the rest. This way, it is possible to establish some QoS guarantees for the traffic. The QoS defined for differentiated services is also known as soft QoS.
- *Guaranteed services.* Guaranteed services take a step further. They are provided by reserving network resources only for chosen traffic flows. Guaranteed services are more QoS-reliable than differentiated services, but they make efficient bandwidth use difficult. The QoS for guaranteed services is also known as hard QoS.

6.1.2 Congestion Management

Congestion is the degradation of network performance due to excessive traffic load. By efficiently managing network resources, it is possible to keep performance with higher loads, but congestion will always occur, sooner or later. So, when delivering services with QoS, one must always deal with congestion, one way or another.

There are two ways to deal with congestion:

1. *Congestion control* is a set of mechanisms to deal with congestion once it has been detected in a switch, router or network. These mechanisms basically consist of discarding elements. The question is: which packets to discard first?
2. *Congestion avoidance* is a set of mechanisms to deal with congestion before it

happens. There are two types of congestion avoidance techniques:

- Admission control operates only at the provider network edge nodes, ensuring that the incoming traffic does not exceed the transmission resources of the network.
- Resource management is used to control the amount of resources being used in the network.

Congestion avoidance, and especially traffic admission, checks the properties of the subscriber traffic entering the provider network. These properties may include the average bit rate allowed in order to enter the network, but other parameters are used as well. For example, a network provider may choose to limit the amount of uploaded or downloaded data. Bandwidth profiles are used to specify the subscriber traffic, and the packets that meet the bandwidth profile are called conforming packets.

Non-conforming packets can be delayed, sent without any QoS commitment, or simply dropped. There is a contract between the subscriber and the service provider that specifies the QoS, the bandwidth profile, and how to deal with the traffic that falls outside the bandwidth profile. This contract is known as the *Service-Level Agreement* (SLA).

6.2 End-to-End Performance Parameters

The first step in offering QoS is to find a set of parameters to quantify and compare the performance of the network. QoS is provided by the network infrastructure, but experienced by the users. This is the reason why QoS is specified by means of end-to-end parameters. Some of the parameters that can be used are end-to-end delay and *Bit Error Ratio* (BER).

The *Internet Protocol Performance Metrics* (IPPM) is an IETF working group that defines standard metrics to evaluate network performance. These metrics are intended for network dimensioning and network equipment design, as well as for QoS and SLA verification. Regarding the latter, there are at least four critical metrics to define:

- Delay
- Delay variation
- Loss
- Bandwidth

The IPPM working group has released *Requests for Comments* (RFC) that deal with these parameters. Other metrics defined by the IPPM include connectivity and packet reordering.

Another goal of the IPPM working group is to define measurement protocols that enable cooperation between measurement equipment. The *One-Way Active Measurement Protocol* (OWAMP) and the *Two-Way Active Measurement Protocol* (TWAMP) have been drafted for this purpose.

The IPPM defines atomic metrics: parameters measured from a minimum amount of data. For example, delay is defined for one single packet. But usually, the information given by a measurement performed in one instance only is not enough to describe the entity under test. A set of related atomic measurements forms a sample measurement. The samples in a sample measurement can be all the possible atomic measurements within the measurement interval, or a more reduced set of atomic measurements obtained at instants chosen statistically over a longer period of time. Finally, there are statistic measurements that summarize the information given by a sample measurement by means of an averaging process.

6.2.1 One-way Delay

The end-to-end *one-way delay* experienced by a packet when it crosses a path in a network is the time it takes to deliver the packet from source to destination. This delay is the sum of delays on each link and node crossed by the packet (see Figure 6.2).

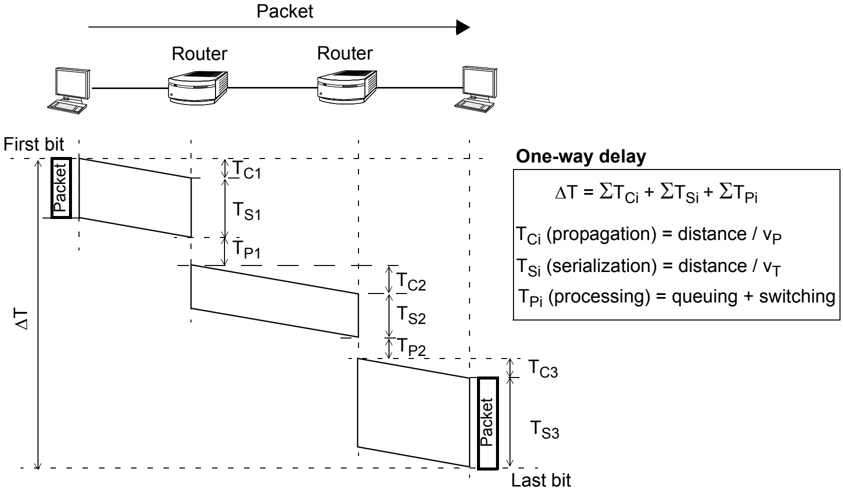

Packet
Router
Router
First bit
Packet
T_{C1}
T_{S1}
T_{P1}
T_{C2}
T_{S2}
T_{P2}
T_{C3}
T_{S3}
Packet
Last bit
ΔT
One-way delay
$\Delta T = \Sigma T_{Ci} + \Sigma T_{Si} + \Sigma T_{Pi}$
T_{Ci} (propagation) = distance / v_P
T_{Si} (serialization) = distance / v_T
T_{Pi} (processing) = queuing + switching

Figure 6.2 One-way delay is the sum of delays on each link and node crossed by a frame.

This definition of one-way delay is atomic, because it is based on a single packet. It is normally more useful to calculate average values of one-way packet delay during a time interval.

The IPPM addresses the definition and measurement of one-way delay in RFC 2679. This RFC deals with IP networks, but the concepts stated there can easily be extended to other packet-switching technologies.

The *Round Trip Delay* (RTD), or latency, is a parameter related to one-way delay. It is the delay of a packet on its way from the source to the destination and back. RTD is easier to evaluate than other delay parameters, because it can be measured from one end with a single device. Packet timestamping is not required, but a marking mechanism of some kind is needed for packet recognition. The best-known RTD tool is Ping. This tool sends *Internet Control Message Protocol* (ICMP) echo request messages to a remote host, and receive ICMP echo replay messages from the same host.

There are three types of one-way delay:

- *Processing delay* is the time needed by the switch to choose the output interface when a packet is received (switching delay), and the time that this packet has to wait in a queue when the transmission interface is busy (queuing delay). The switching delay is often much shorter than the queuing delay, and it is sometimes ignored.
- *Serialization delay* is the delay between the transmission time of the first and the last bit of a packet. It depends on the size of the packet. Some intermediate nodes may need to receive the full packet before forwarding it to an output port, but this is not always so. For example, some Ethernet switches are able to forward frames as soon as the destination address is known.
- *Propagation delay* is the delay between the time the last bit is transmitted at the transmitting node and received at the receiving node. It is constant, and it depends on the physical properties of the transmission channel. It is also proportional to the distance between the transmitter and the receiver.

6.2.1.1 Measurement Methodology

The accuracy of measuring one-way delay is important, because various error sources need to be controlled. In the past, one-way delay was estimated to be half of the RTD result. In most of the cases today, this is no longer acceptable, due to asymmetry in the transmission path.

To understand the reasons that can compromise the accuracy of a one-way delay measurement, it is first necessary to know what are the typical parameters of the clocks needed to make the measurement:

- The *offset* between two clocks is the difference between the time given by these clocks.
- The *accuracy* of a time source is the offset of that source compared to the UTC.
- The *skew* gives a change rate of a clock's offset or accuracy over time. It can be said, for example, that the skew of a clock is 10 ms per day.
- The *drift* is the rate of change of the skew over time. It depends on the temperature.
- The *resolution* is the minimum time difference that can be distinguished with a clock. In other words, it is the duration of a clock tick.

It is important to note that these parameters are based on the time given by a clock. It is possible to define a different set of parameters based on the frequency instead of time. For example, a frequency offset between two clocks is related to the time skew between them.

The errors and uncertainties in one-way delay measurements are the following:

- *Synchronization errors between source and destination clocks*: A one-way delay measurement requires the transmitter and the receiver to be synchronized to maintain the time offset under control. In IP networks, this can be achieved by using the *Network Time Protocol* (NTP). The performance of this protocol is quite unpredictable, but it is accepted that an error in an NTP-based one-way delay measurement is a few milliseconds at worst. This can be accepted when measuring delays of hundreds of milliseconds. The *Global Positioning System* (GPS) timing or SDH-derived timing offer performance levels several orders of magnitude better than NTP, but GPS is expensive and requires installing external antennas – and SDH is not always available.
- *Errors related to wire times and host times*: Bits appear and are detected in the transmission channel at wire times. On the other hand, one-way delay measurements involve packet timestamping. But packets are timestamped at host times that might not be the same as wire times. The difference between these two times is usually very difficult to predict.
- *Uncertainty related to the resolution of measurement clocks*.

6.2.2 One-way Delay Variation

The *one-way delay variation* of two consecutively transmitted packets is the one-way delay experienced by the last transmitted packet, minus the one-way delay of the first packet (see Figure 6.3). This is defined by the IPPM in RFC 3393. Accord to the definition, involving only two packets, one-way delay variation is atomic. More useful results are obtained by averaging several atomic measurements.

There are many parameters related to one-way delay variation:

- *Jitter* is a term widely used in classic telephony. If this term is used for packet-switched networks, its meaning is similar to one-way delay variation. However, while one-way delay variation may be a negative number, packet jitter is normally supposed to be non-negative, and therefore it is often referred to as the absolute value of one-way delay variation.
- *One-way delay variance* is another possible parameter for delay variation. It is a non-negative number and a statistical value. So, unlike one-way delay variation, one-way delay variance is the result of an averaging process.
- *Peak-to-peak delay variation* is the difference between the maximum and minimum values of a sequence of delay variation samples.
- The *Interarrival time* between two consecutively received packets is the difference between the arrival time of the last and the first packet. Timestamping or sequencing is not required to measure interarrival time, which makes this parameter easier to measure than one-way delay variation. This metric is useful to evaluate the impact of variable delay in packet periodicity but it is not always a good metric for delay variation.

Delay variation is present in both circuit- and packet-switched networks. In circuit-switched networks, delay variation has to do with justification events due to the skew between different clocks. In packet-switched networks, the main sources of delay variation are: variable queuing times in the intermediate network elements, variable serialization and processing time of packets with variable length, and variable route delay when the network implements load-balancing techniques to improve utilization. This makes delay variation in packet networks more remarkable and unpredictable than in circuit switched networks.

6.2.2.1 Measurement Methodology

Evaluating one-way delay variation is easier than measuring one-way delay, because the traffic generator and analyzer do not need to be synchronized. The offset between clocks is cancelled, because one-way delay variation is the difference between two delays. Skew usually changes slowly with time and does not affect the results.

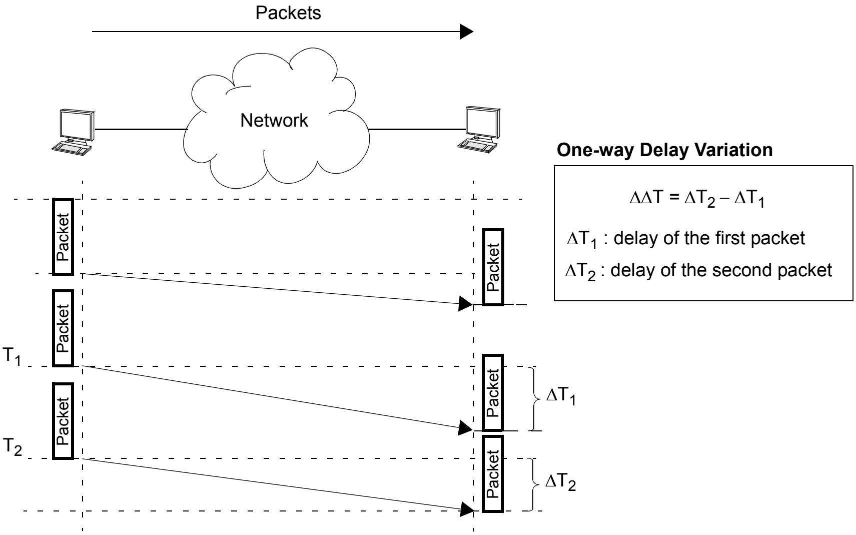


Figure 6.3 One-way delay variation: measurement and impact on data periodicity

Although high accuracy is not necessary, one-way delay must be evaluated before one-way delay variation. This is the reason why timestamping of probe packets is required.

The error sources for one-way delay derive from wire and host times and from the uncertainty caused by the limited resolution of the clocks used. The uncertainty of one-way delay is twice as high as that of one-way delay.

6.2.3 Packet Loss

A packet is said to be lost if it does not arrive to its destination. It can be considered that packets that contain errors or arrive too late are also lost. “Too late” may have different meanings for different applications. In real-time audio and video applications, packets received later than they should have been received are not useful, but it is possible to be less restrictive in many data applications.

The IPPM addresses packet loss evaluation in RFC 2640. The definition given there is based on an application-dependent timeout. The IPPM definition involves only a single packet, and therefore the packet can only take two values: “0” when it is successfully received before the timeout, and “1” when it is lost.

Making packet loss dependent of timeouts may cause problems in practice, because most of the times it requires packet timestamping and synchronization between the analyzer and the generator. An exception of this is the round-trip packet loss measurement. In this case timestamping and synchronization are avoided, and only packet sequencing is needed. Ping is a well-known tool for measuring round-trip packet loss.

Nevertheless, there is an alternative to synchronized one-way packet loss, if packets form a part of a sequenced stream: A packet is considered to be lost if a packet is received with a higher sequence number than expected. For networks that allow packet reordering, it may be necessary to let several packets pass with higher sequence numbers before considering a packet to be lost.

A single packet loss event is not meaningful enough to describe the properties of the path between the transmitter and the receiver. There are statistical metrics that give more accurate descriptions. For example, packet loss is often expressed as a ratio of lost and transmitted packets:

Packet loss may occur when transmission errors are registered, but the main reason behind these events is network congestion. Intermediate nodes react to high traffic load conditions by dropping packets and thus generating packet loss. Congestion tends to group loss events, and this harms voice and video decoders that are optimized to work with uniformly distributed loss events.

Loss distance and loss period are metrics that give information on the distribution of loss events (see Figure 6.4).

- *Loss distance* is the difference in the sequence numbers of two consecutively lost packets, separated or not by received packets.
- *Loss period* is the number of packets in a group where all the packets have been lost.

These atomic parameters make it possible to collect statistics that describe the distribution of packet loss events between the transmitter and the receiver.

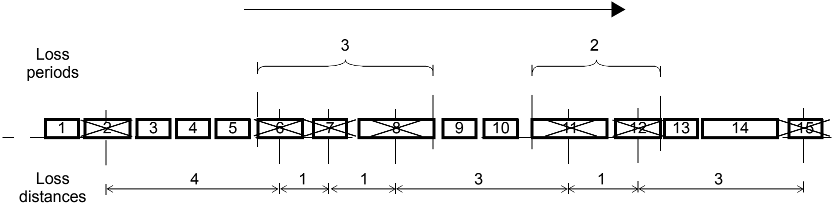


Figure 6.4 One-way delay variation: measurement and impact on data periodicity

6.2.4 Bandwidth

Bandwidth is a measure of the ability of a link or a network to transfer information during a given period of time. There are different metrics for bandwidth:

- Bandwidth *capacity* defines the maximum rate at which packets can be transmitted by the link.
- *Available bandwidth* measures the bit rate a link or a network can accept for transmission at a given moment.

Bandwidth capacity does not usually change with time, but bandwidth availability depends on the traffic load, and therefore it varies in time. If R_i is the capacity of a link (i), A_i its available bandwidth, and u_i a number between 0 and 1 that measures the utilization the link, then:

$$\Delta_i = R_i(1-u_i) \qquad \textit{eq. 6.1}$$

Capacity and available bandwidth can be defined for links, or for entire transmission paths composed of various links. The capacity of an end-to-end path is the smallest link capacity of the links that make up the path (see Figure 6.5). The link that limits the end-to-end capacity is called a *narrow link*. A similar definition can be given for the available bandwidth of an end-to-end path. In this case, the limiting link is called *tight link*. Both tight and narrow links are sometimes called *bottleneck links*. This term may be confusing, because it does not make distinction between capacity and available bandwidth.

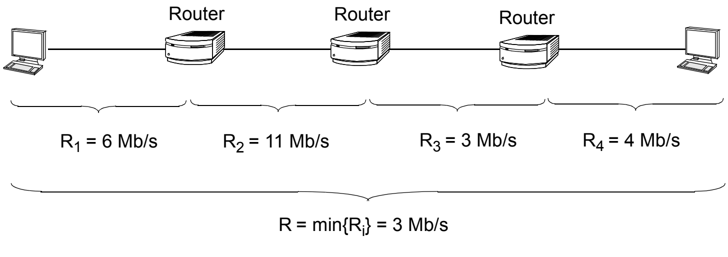


Figure 6.5 The capacity of an end-to-end path is the smallest link capacity of the links that make up the path.

For QoS, the most important bandwidth metric is the available end-to-end capacity, because only end-to-end parameters are relevant when evaluating a service. In RFC 3118, the IPPM defines a very similar metric, namely *Bulk Transfer Capac-*

ity (BTC). This measures the ability of a network to transfer information over a single congestion-aware transport connection, such as the *Transfer Control Protocol* (TCP).

6.2.4.1 Measurement Methodology

Measuring and estimating bandwidth are challenging tasks. The algorithms available to do this show the connection between bandwidth estimation and delay measurement. The most simple method for measuring bandwidth is the *Variable Packet Size* (VPS) model. This method assumes store-and-forward retransmission in intermediate devices, as well as linear dependency of delay with packet length.

The delay experienced by a single packet in a lightly loaded packet network in the first l links can be approximated in the following way:

$$\Delta T_l = \sum_{i=0}^{l-1}\left(\frac{s}{R_i}+d_i\right) = s\sum_{i=0}^{l-1}\frac{1}{R_i}+D = sk_l+D \qquad \textit{eq. 6.2}$$

Here, d_i is the transmission delay of the link (i), and s is the packet size. This is the equation of a line with slope, k_i. By delivering packets with different lengths to the destination and to the intermediate nodes, it is possible to collect enough data to calculate link bandwidth, R_i.

The VPS model has many inconveniences. One of them is that it does not consider queuing delay in intermediate nodes. Many network elements have shown significant deviation from the behavior expected by the algorithm, and this is the reason why there are many versions of this basic method, and other different algorithms have been proposed to measure bandwidth.

6.3 MARKING

Marking means classifying packets by means of dedicated header fields, to identify traffic flows or classes.

Traffic marking is closely related to traffic and flow identification. Network elements can efficiently mark packets and process marked packets. However, identifying unmarked traffic flows can be difficult and time-consuming, if properties such as the transport layer port, *Multipurpose Internet Mail Extension* (MIME) type or *Uniform Resource Locator* (URL) are taken into account for classification.

Sometimes, marking is associated with traffic admission processes. Marks can be set by the subscriber, service provider, or both, for different purposes:

- *To identify the CoS carried within the payload.* For example, VoIP and data traffic have different QoS requirements, and they may be marked with dedicated labels to allow for a customized treatment by the network.
- *To deal with non-conforming packets.* Sometimes, traffic that does not meet the bandwidth profile is not dropped or delayed, but simply delivered without any QoS guarantees. If this is the case, the provider edge node may need to place the correct label to these packets.
- *To enable a customized per-subscriber service provision.* Different customers may have different requirements and SLAs. Marking each customer's packets with a different label makes it easier to provision their customized services.

Traffic marking also has to do with congestion control. A network element that is experiencing congestion in a privileged traffic class may choose to re-classify a part of the traffic to a non-privileged class to free up resources.

Traffic classification can be microscopic (per flow) or macroscopic (per class), but marking is used for both.

6.3.1 Traffic Flows

A *traffic flow* is a unidirectional sequence of packets that are related between them and have the same QoS requirements. It is commonly understood that a traffic flow results from the activity of a single user. For example, the set of packets that make a *Hypertext Transfer Protocol* (HTTP) request to a web server is an example of a flow. Another example is the sequence of packetized voice samples from a *Voice over IP* (VoIP) application coming from one user. A video conference with both video and audio can be made up of several traffic flows (one for audio and one for video, for instance) and they may have different QoS requirements. Signaling packets associated with this type of multimedia communications usually have special QoS requirements, and they belong to different flows than data packets.

Every flow in a network may have its own QoS requirements, and one possible approach to QoS provisioning is to share network resources on a per-flow basis, assigning every flow its own bandwidth, memory and processor time in network elements. If this solution is implemented, flow identification is a problem that needs to be solved. IP packets include the source and destination address, and they can be used for this purpose. But this is usually not enough, because IP addresses identify communicating parties, not applications. The transmission port and the MIME type include more details that can help in flow identification, but yet a better solution is to mark flows according to their specific QoS requirements at layer 3 or layer 2. The IP precedence or the IEEE 802.1p Ethernet CoS fields provide a way to do this.

6.3.2 Traffic Classes

There are traffic flows with similar QoS requirements. All VoIP flows are delay-sensitive, and all web transactions are sensible to transmission errors and losses. These sets of microscopic flows can be grouped into larger entities called *classes*. Traffic classes receive the same treatment in the provider network, and each class may use dedicated resources in the network elements they pass through. It is often a good idea to keep traffic coming from the same application within the same class.

Table 6.1
Standard QoS requirements for different applications

Traffic type	*Bandwidth*	*Packet loss (max)*	*Delay (max)*	*Jitter (max)*
Interactive voice (G.711)	12 ~ 106 kb/s	1 %	150 ms	30 ms
Streamed video (MPEG-4)	0.005 ~ 10 Mb/s	2 %	5000 ms	insensitive
Streamed audio (MP3)	32 ~ 320 kb/s	2 %	5000 ms	insensitive
Data	variable	sensitive	insensitive	insensitive

Traffic classes are marked with CoS labels that enable efficient processing. The main packet-based network technologies provide header bytes for marking traffic classes. These technologies include IP, ATM, Frame Relay, MPLS and Ethernet.

By classifying traffic, network resource management can be improved, keeping usage low for critical traffic. Congestion can be controlled so that it will only affect non-critical traffic classes. Another advantage of classification is that it makes it possible to use intelligent packet scheduling in switches and routers. CoS labels are useful when deciding the transmission order for those packets that are queuing for transmission.

6.4 Scheduling

A *scheduler* is a logical component of a network element that chooses the next packet to be sent to an output interface from among all candidates waiting in queues.

The decision made by the scheduler can be based on the waiting time or any other characteristic of the traffic. When the scheduler takes traffic flows or class-related characteristics into account it becomes a key component of traffic differentiation and customized QoS provision.

There are some properties that a good scheduler should always have:

- It may be necessary to provide some *guarantees* to the scheduled traffic, or at least to a part of it. For example, some flows may require the queuing time to be shorter than a predefined interval.
- *Fair bandwidth sharing* between some or all of the flows. Privileged flows should not be allowed to consume all the bandwidth most of the time.
- *Processing time* in network elements must be managed carefully. Simple and easy-to-implement schedulers are preferred.

6.4.1 First In, First Out Scheduler

The *First In, First Out* (FIFO) scheduler provides basic store-and-forward capability. It only considers the order of arrival of the incoming packets. Apart from this, no other characteristics are taken into account. This is a very simple approach, but it has some important drawbacks:

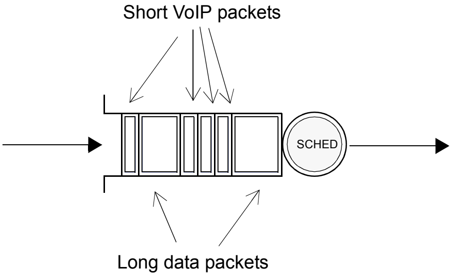


Figure 6.6 VoIP packets have to wait in the FIFO scheduler for the data packets to be served. This disturbs the end-to-end QoS of the VoIP application, even if there are just a few data packets.

- The QoS-demanding multimedia traffic has to stay in the same queue with bursty data traffic. A short VoIP packet that is in the queue after a long data packet must wait until the whole data packet has been transmitted. This makes queuing delays highly dependent on the bandwidth profile of the bursty data source.
- There is no mechanism to control how the bandwidth is shared between the flows. Those flows that use more bandwidth tend to get most of the resources managed by the scheduler.

6.4.2 Round Robin Scheduler

The *Round Robin* (RR) scheduler treats traffic quite fairly, with low computational cost. The Round Robin algorithm polls several queues sequentially. Empty queues are skipped. Usually, packets from different flows or classes are stored in separate queues. The available bandwidth is shared fairly between these flows or classes, if all packets have the same length. Otherwise, flows with bigger average packet size are favored.

The basic RR scheduler assigns the same bandwidth to every flow in those technologies that use fixed-length packets, but it can be modified to work with flows that use customizable bandwidths. The modified RR scheduler is called *Weighted Round Robin* (WRR), and it calculates a service schedule based on shared coefficients assigned to each queue. This means that packets in a queue with size that doubles the size of other queues are served twice as fast.

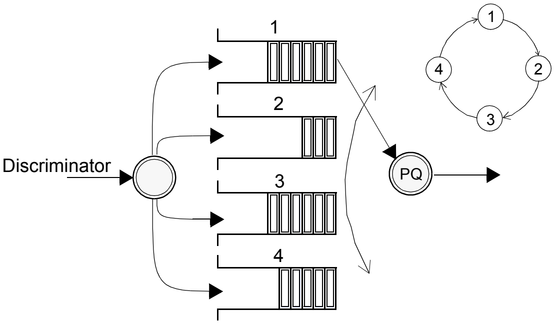


Figure 6.7 The Round Robin algorithm polls several queues sequentially, ignoring empty queues. Traffic flows are treated fairly.

The inconvenience of the RR and WRR algorithms is that they do not consider the length of the packets. In fact, they are not useful at all for technologies with variable packet lengths. There is a modified version of the basic RR algorithm called *Deficit Round Robin* (DRR) that deals with unpredictable-length packets. The idea behind the DRR is to keep track of the deficits experienced by a flow in one round, and compensate for these deficits in the next round. To do this, the DRR algorithm assigns a value called *quantum* for every queue. The quantum value is proportional to the level of service assigned for every queue. Additionally, there is a deficit counter on each queue, incremented by a quantum at every round. Packets can only be scheduled if their length is smaller or equal to the deficit counter. Once a packet has been served, the deficit counter is decremented by the packet length, and set to zero when empty queues are found.

6.4.3 Weighted Fair Queuing

The *Weighted Fair Queuing* (WFQ) scheduler is an approximation of a theoretical model known as *Generalized Processor Sharing* (GPS). A GPS scheduler serves traffic queues as if the traffic were a fluid flow. A hypothetical GPS scheduler has several interesting properties: it creates bit-wise fairness for the transmitted traffic, makes it possible to configure a specific bandwidth for each traffic flow, and gives delay and delay variation guarantees. Unfortunately, the GPS scheduler requires packets with infinitesimal size, which is why it cannot be implemented.

The WRR and DRR schedulers are also approximations to the GPS scheduler but they do not treat traffic flows fairly in short time scales or when there are many queues. The WFQ gives better results in these situations, but it has a higher processing load, and it is more expensive to implement.

6.4.4 Priority Scheduler

The *Priority scheduler* ensures that the more important the traffic is, the quicker it is handled. Packets with low priority are not delivered, if there are packets with higher priority waiting to be sent.

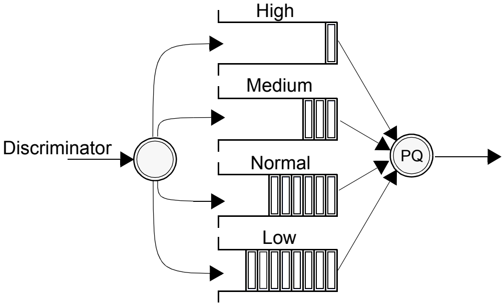


Figure 6.8 The Priority scheduler does not send low-priority traffic if there are packets with higher priority waiting to be sent.

The Priority scheduler is easy to implement, and it does not use up a large amount of resources, but giving absolute preference to high-priority traffic can be inconvenient in practice, because lower-priority traffic may have to wait for a long time before it can be sent, and this is not always desirable. When the available bandwidth needs to be distributed fairly, other, more complex schedulers (such as DRR or WFQ) must be used.

6.5 Congestion Avoidance

Congestion avoidance is used to prevent congestion before it occurs. This is done by controlling the amount of traffic in the network as carefully as possible.

Congestion avoidance techniques use open-loop and closed-loop flow control mechanisms. Closed-loop mechanisms rely on feedback messages given by intermediate or end-user equipment to prevent congestion. A good example of closed-loop congestion avoidance is the window-based mechanism provided by TCP in IP networks.

Closed-loop congestion avoidance is well-suited to data networks, but it is slow with multimedia traffic. Open-loop congestion avoidance is the best option for the latter.

There are two open-loop congestion avoidance techniques:

1. *Admission Control* filters traffic by discarding, delaying or re-classifying packets or flows, to ensure that the transmission resources of the network are not exceeded.
2. *Resource Management* functions reserve resources in the network before starting transmission, and free up resources when transmission is over.

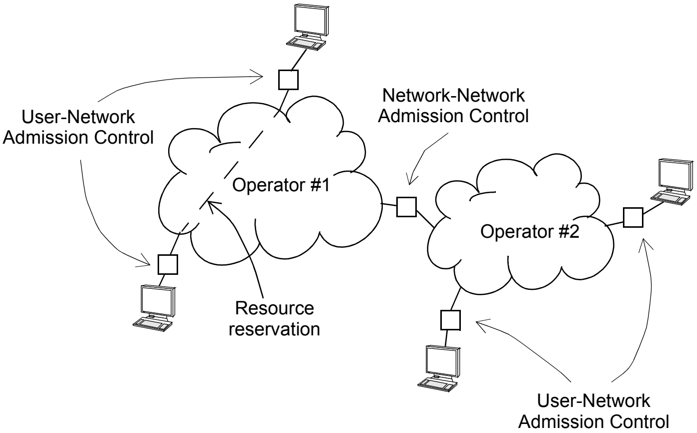


Figure 6.9 Resource management operates within the network, and admission control at the boundaries of the network, but admission control can also be used to split a large network, or to separate networks that belong to different operators.

Resource management is implemented by means of protocols that operate when connection parameters are negotiated or re-negotiated, or when a connection is released. For non-connection-oriented technologies, such as IP, resources are reserved and freed up on a per-flow basis. These protocols are understood by the switches and routers that make up the network, and new connections or flows are accepted by them when there are resources available. Otherwise new connections are rejected.

Admission control is often used at the boundaries of the network, but it can be applied to intermediate network elements as well. When admission control is applied to intermediate network elements, it is generally called *traffic shaping* or *traffic policing*. Shaping or policing can, for example, be used to separate networks that belong to different operators. Resource management calls for compatibility between reservation protocols in all the network elements used, and operators have to make an agreement on how to reserve resources for each other. These problems may be solved by implementing the necessary shaping and policing functions.

6.5.1 Admission Control

There are two problems related to admission control. First, how to find an algorithm that can decide if a particular packet is conformant or not, and secondly, what to do with non-conformant packets? Most of the admission control algorithms are based on the token bucket system (see Paragraph 6.5.1.1).

There are different types of filters that can help to classify non-conformant packets, and each of them have different effects on the traffic (see Figure 6.10):

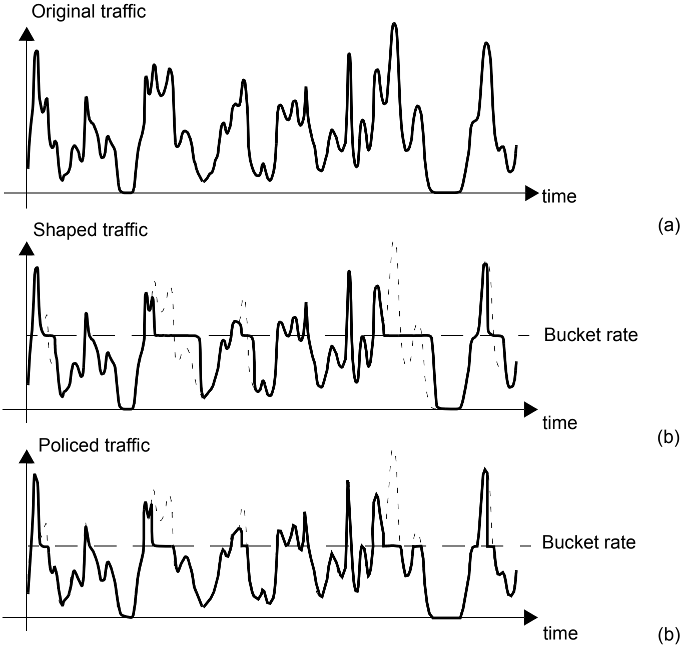


Figure 6.10 Shaping and policing of user traffic. (a) When traffic is shaped, no packets are dropped, but some of them may be delayed. (b) When traffic is policed, it is never delayed, but some packets may be dropped.

- *Policers* are filters that discard all non-conformant packets. Because packets are dropped, the original information is not fully conserved, but timing is not affected, so, packets are not delayed. Policers are well-suited to those error-tolerant applications that have strict timing constraints, for example VoIP or some interactive video applications.
- *Shapers* work much the same way as policers, but they do not discard packets. Non-conformant traffic is buffered and delayed until it can be sent without violating the SLA agreement or compromising network resources. Shapers conserve all the information that was sent, but they modify timing, so they may cause problems for real-time and interactive communications.
- *Markers* can be used to deal with non-conformant packets (see Paragraph 6.3). Instead of dropping or delaying non-conformant packets, they are delivered with low priority or "best effort".

Policers, shapers and markers can be combined to obtain more complex filters that have very interesting properties. For example, there could be an admission control filter that acts as a policer in situations where the SLA agreement is strongly violated, but in the case of eventual or non-persistent violations, the filter would send non-compliant packets – without any QoS guarantees, of course.

6.5.1.1 The Token Bucket

Most of the admission control algorithms are based on the *token bucket* system. These algorithms include a *conceptual bucket*, of size S, that is filled with tokens at a configurable rate, R. Each packet needs to take a token from the bucket before it can be sent. If there are tokens available, the packet can be delivered, but if the bucket is empty, the packet needs to be treated somehow, and the treatment it gets depends on the type of filter that is being used: Policers drop packets when there are no tokens, markers re-classify them, and shapers store them in a buffer until more tokens are available (see Figure 6.11). A bucket can store S tokens at maximum. When the bucket is full, it cannot receive any more tokens, so any extra tokens are discarded.

The token bucket algorithm is simple, and it clearly separates conformant and non-conformant packets. It sets limits for the amount of data that enters the network during a specified time interval. Perhaps the most interesting property of this system is that it can be used to make sure that the delay for those flows that are shaped and policed by this algorithm stays within previously defined limits. This way, it possible to provision guaranteed services in packet-switched networks (see Paragraph 7.2.1.1).

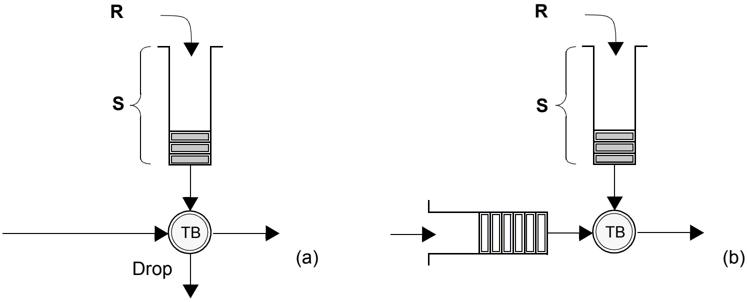


Figure 6.11 Token buckets used in admission control: (a) Token bucket working as a policer. (b) Token bucket working as a shaper.

6.5.1.2 Single-Rate Three-Color Marker

The *single-rate Three-Color Marker* (srTCM) policer is obtained by chaining two simple token bucket policers (see Figure 6.12). Tokens fill the main bucket until they reach the capacity given by the *Committed Burst Size* (CBS) parameter, at a rate given by the *Committed Information Rate* (CIR) parameter. Any extra tokens in the main bucket are not ignored, but used to fill a secondary bucket until they reach the capacity given by the *Excess Burst Size* (EBS) parameter.

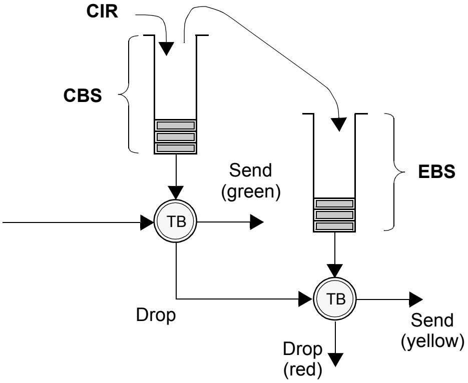


Figure 6.12 Single-rate three-color marker policer

The traffic that passes through the first bucket (*green traffic*) is delivered with the QoS agreed with the service provider, but any traffic that passes through the secondary bucket (*yellow traffic*) is usually re-classified and delivered as best-effort traffic, or it is given a low priority. Non-conformant traffic (*red traffic*) is dropped.

The srTCM can be considered as a more sophisticated version of the simple token bucket policer. Some non-conformant traffic that would otherwise be dropped can be recovered by using srTCM.

6.5.1.3 Two-Rate Three-Color Marker

The *two-rate Three-Color Marker* (trTCM) algorithm is a modified version of the srTCM that includes a new configuration parameter, the *Excess Information Rate* (EIR). The difference between the two is that with trTCM, the excess tokens from the main bucket are not used to fill the secondary bucket, but the secondary bucket is filled by a new token flow at a rate given by the EIR parameter (see Figure 6.13).

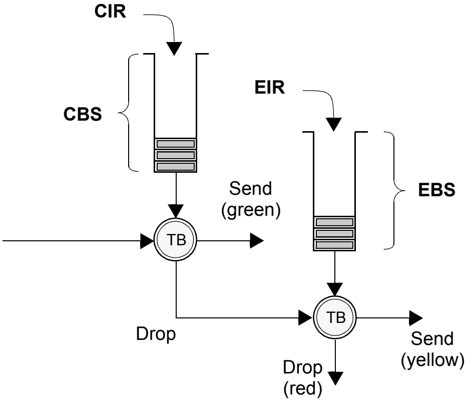


Figure 6.13 Two- rate three-color marker policer

In practice, the only important difference is that the trTCM allows for a more accurate control over yellow traffic by means of the EIR parameter.

6.5.2 Resource Management

The classical circuit-switched networks can accept a limited number of simultaneous calls. If this number is surpassed, the network becomes unavailable for new callers. The existing callers do not notice any change in the quality of service. When the network is unavailable for new callers, it is said that it is *blocked*.

In the traditional packet-switched networks, the situation is quite different. When the traffic load is small, these networks remain predictable and provide good performance. But QoS degrades quickly when there is more traffic load.

For many audio and video applications, continuous performance degradation is not acceptable. For these applications, it would be better to have the behavior of a circuit-switched network and the flexibility of a packet-switched network.

Circuit-switched networks can establish end-to-end connections, although some packet-switching technologies can do the same. But while circuit-switching technologies provide physical end-to-end connections, the circuits provided by packet-switching technologies are virtual (Virtual Circuits, VC): they try to emulate the properties of physical circuits.

Those technologies that are based on VCs, for example ATM, can potentially provide the same level of service as any other circuit-switched network, while maintaining high flexibility. VCs do have some drawbacks as well, because they need:

- *A resource-management protocol* to establish and release connections. Usually these signaling protocols are complex and bandwidth-consuming.
- *To store flow status information* in intermediate nodes. This means that it is necessary to add complexity to network elements by adding more memory and processing time to manage connection information.
- *To waste time establishing connections*. The time wasted in establishing a connection means extra delay.

IP networks are not circuit-oriented due to design reasons. These networks deliver packets, and all the information needed for these packets to arrive to their destination is included in the packet header. These packets are called datagrams. Packet-switched networks based on datagrams are simple and scalable, but they are not the best possible solution to transport voice and video.

When the users of an IP network need connection-oriented communications, they use TCP. It provides packet re-ordering, error detection, re-transmission of errored packets, end-to-end flow control and other features, but it cannot establish any VCs in the network. TCP connections only reside in the end user equipment. The network is not aware of them. Connection-oriented communications across the IP network would be desirable, but this is not possible without changing the fundamental design of the IP protocol. There are currently two main options to offer circuit-oriented transmission over IP:

1. *Reservation Protocol (RSVP)*. The RSVP is the most important of all the resource management protocols proposed for IP. It is an important component of the *Integrated Services* (IS) architecture proposed for IP networks. This architecture actually turns IP into a connection-oriented technology. To be efficient, the RSVP needs to be supported by all the network elements, and not only by the end user equipment. Both RSVP and IS call for a new generation of

IP routers.

2. *MPLS*. MPLS is a switching technology based on labels carried between the layer-2 and layer-3 headers that speed up IP datagram switching. MPLS can be used for QoS provisioning in IP networks. One of the reasons for this is that MPLS supports a special type of connections called *Label-Switched Paths* (LSP). The LSP setup and tear-down relies on a resource management protocol, usually the *Label Distribution Protocol* (LDP), but RSVP with the appropriate extension for MPLS can be used as well.

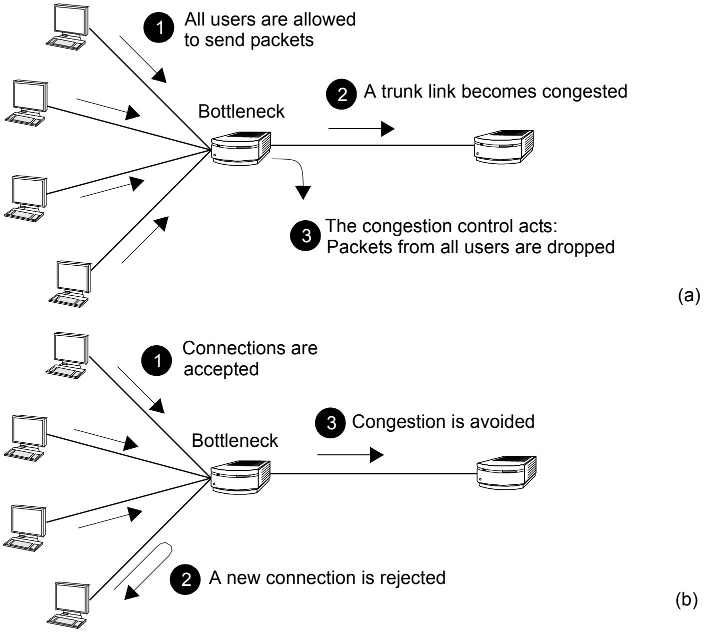

1 All users are allowed to send packets
2 A trunk link becomes congested
Bottleneck
3 The congestion control acts: Packets from all users are dropped
(a)
1 Connections are accepted
3 Congestion is avoided
Bottleneck
2 A new connection is rejected
(b)

Figure 6.14 How resource management acts: (a) Without resource management, all users experience degradation on their applications whenever there is congestion in the network. (b) If congestion management is used, only some subscribers are not allowed to send data, but the others are not affected.

RSVP and MPLS have one inconvenience in common: they rely on a complex resource management protocol. How to manage IP connections without a protocol of this type is still an open question.

6.6 Congestion Control and Recovery

Congestion control is the set of techniques that deal with congestion once it has been detected in the network, and its aim is to minimize damage on communications.

In a packet-switched network, a link becomes congested whenever the amount of injected traffic exceeds its transmission capacity. The first consequence of this is that excess packets need to be buffered. This may potentially damage the QoS of real-time multimedia communications. If congestion is severe, buffers may become overloaded and cause packet loss, damaging both data and multimedia communications. Correct packet marking and scheduling help to deal with any damage caused by queuing delays. From this point of view, traffic differentiation is related to congestion control. To deal with the effects of packet loss, it is necessary to implement good *Packet Dropping Policies* (PDP) for network elements. The ideal features of a good PDP are:

- *Efficiency.* A good PDP should keep network utilization high. Dropping fragmented packets could reduce efficiency. ATM fragments IP datagrams before transmitting them, but there is no mechanism to recover a single lost of errored ATM cell. This makes it necessary to request the transmission of the entire IP datagram at the destination. Fragments that arrive without errors are not usable, but they waste transmission resources. To avoid this problem, the PDP for ATM cells must be aware of the fragmentation of IP packets and drop cells on a per-IP-packet basis rather than on a per-ATM-cell basis. Similar problems occur at flow level. Some flows may not be usable, if some packets are lost, so it makes sense to drop entire flows. For the particular case of TCP, a PDP that does not consider TCP connections may cause network underutilization. A single packet loss in a TCP flow would cause the activation of the TCP congestion avoidance mechanism, and TCP would reduce the transmission rate for that connection. If this happens to many TCP connections at the same time, it may lead to underutilization of the network (global synchronization problem). Following the same reasoning, it would make sense to first discard packets that have traversed a smaller number of hops.
- *Fairness.* There should never be more privileged flows than those explicitly allowed by the service provider. Under the same circumstances, all flows should have the same statistical chances of being transmitted without packet loss. If "unfairness" is decided by the service provider, it should be possible to control the level of "unfairness".
- *Simplicity.* As network elements usually have to switch traffic at high rates, it is necessary to keep PDPs as simple as possible. There is a trade-off between efficiency and simplicity. The more efficient and intelligent a PDP is, the more complex it is.
- *Scalability.* PDPs must be prepared for rapid growth of current networks. Scalability tends to be a problem for those PDPs that store connection information.

6.6.1 Drop Tail

The *Drop Tail* (DT) is the simplest, but not always the best PDP. It accepts packets whenever there is buffer space available, and when there is no space, it drops all incoming packets regardless of their type, priority or any other property.

The DT PDP is very scalable, because it is simple, but it has some efficiency and fairness problems. Although DT can be accepted in networks with very light loads, it is not recommended when high utilization level is expected. In fact, it has been demonstrated that in this case the DT can entirely shut down the system.

The DT is biased against flows with small amount of traffic, and a few high-rate flows tend to get most of the system resources. The DT PDP does not take packet fragmentation or packet flows into account. Long packets have a smaller probability to pass trough the policer than short messages when they are fragmented. Furthermore, it has been shown that the TCP protocol exhibits low performance when DT is used.

6.6.2 Partial Packet Discard

The *Partial Packet Discard* (PPD) PDP has been proposed for ATM networks, to improve the performance of the DT. The improvement consists in that whenever the PPD has to drop a packet fragment, it also drops all subsequent fragments belonging to the same packet. This saves storage space for other packets, but some fragments that are not usable are still forwarded. On average, one half of the unusable fragments are delivered, which means that it is possible to further improve this PDP.

The PPD needs to store information about packets with previously discarded fragments. This status information adds complexity to this PDP. This is not a big problem for ATM, because the connection table of ATM switches can easily be extended to store the information needed by the PPD.

6.6.3 Early Packet Discard

The *Early Packet Discard* (EPD) PDP is an alternative improvement for the DT in ATM networks. The buffer with EPD does not accept any more packets when it exceeds a certain configurable threshold. However, fragments from those packets that were already accepted before the threshold was exceeded are queued if there is buffer space available.

The EPD is as complex as the PPD. This PDP needs to store information on all the queued fragments. Performance-wise, the EDP has been demonstrated to be better than PPD in most circumstances. Of course, EPD and PPD can be used at the same time to improve performance even further.

There are many extensions and customizations of the EPD for different purposes; for example, to deal with TCP connections or to improve fairness.

6.6.4 Random Early Detection

The *Random Early Detection* (RED) PDP is designed to work with IP. It does not need to store any flow status information, and it collaborates with the TCP congestion control mechanism.

The RED splits the storage queue into three regions defined by two thresholds, T_{low} and T_{high}. The RED has different behaviour depending on the queue length (see Figure 6.15).

- *Green region*. The RED operates in this region if the queue is shorter than T_{low}. In this case, all incoming packets are accepted.
- *Yellow region*. This region is defined to be between T_{low} and T_{high}. When RED is operating in this region, an incoming packet is discarded randomly with a probability of P_a
- *Red region*. RED works in this region if the queue is longer than T_{high}. In this case, all incoming packets are dropped.

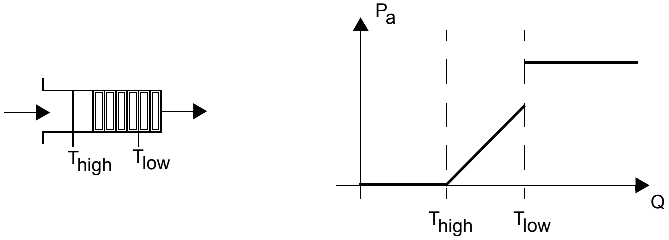


Figure 6.15 Packet drop probability in a queue with RED PDP as a function of the average queue length

Actually, the operating region does not depend directly on the instantaneous value of the queue length. An averaged version, Q, is used for this purpose, to avoid changing the operation region too often. The probability of dropping a packet in the yellow region depends on two factors:

1. *Average queue length*: As the queue length approaches to T_{high}, the dropping probability increases.
2. *Counter*: Every time a packet is queued, the counter is incremented. This counter is reset when a packet is discarded. High counter values increase the probability of packets getting dropped. In other words, the higher the values are, the more likely it is that packets get dropped."

The dropping probability is calculated by means of the following formula:

$$P_a = \frac{P_b}{1 - counter \cdot T_{low}} \qquad eq.\ 6.3$$

Where P_b is:

$$P_b = P_{max} \frac{Q - T_{low}}{T_{high} - T_{low}} \qquad eq.\ 6.4$$

Choosing the right values for T_{low} and T_{high} is important. T_{high} is related to the maximum delay allowed for a packet, and it is especially important in real-time communications. On the other hand, a value for T_{low} that is too small may affect the quality of bursty data traffic.

The RED avoids dropping packets from many TCP connections at the same time, and at the same time helps to solve the problem of "underuse" which is caused by the global synchronization problem (see page 191).

The *Weighted RED* (WRED) is a variation of the RED algorithm that takes the relative priorities of different packets into account. The algorithm starts dropping low-priority packets before dropping any packets with higher priority.

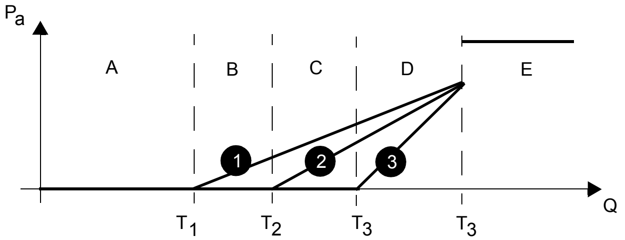


Figure 6.16 A possible implementation of the WRED. When the algorithm is working in region A, all packets are accepted. In region B, packets marked with 1 may be dropped. In region C, packets marked with 2 may be dropped as well. In region D, all three possible classes (1, 2 and 3) can be dropped. Finally, in region E, no new packets are accepted

Chapter 7

QoS Architectures

ATM was the first packet technology that was able to provide custom QoS to subscribers, but most players in the telecommunications market are now abandoning ATM and basing their most innovative QoS solutions on IP.

They have chosen IP, because it has become widely available. However, the legacy IP infrastructure is not QoS-aware, and it can only provide best-effort data services, so the IP technology needs to be modified to meet the needs of the market. The lessons learnt from ATM must not be forgotten: the IP QoS must be as simple and efficient as possible.

It is necessary to find a way to achieve a smooth transition from legacy IP to QoS-aware IP, but the task is not easy. QoS provisioning calls for end-to-end support of dedicated signaling protocols, advanced scheduling techniques, as well as innovative congestion management technologies. IP-based QoS can only be provided in small and medium-size networks – worldwide QoS services over the Internet are not yet possible.

7.1 QoS in ATM Networks

ATM is a versatile technology designed to carry heterogeneous information, including not only data services but also real-time and non-real-time voice and video. In other words, it is designed for a universal converged network.

ATM is a packet-based transmission and switching technology that uses small, fixed-length packets called *cells*. ATM is connection-oriented, which means that during the process of establishing a connection, switches can reserve resources for new traffic in advance. When reserving resources for users, the network can guarantee a certain QoS level. So that ATM switches could reserve enough resources to satisfy the needs of a new connection, the user must specify the bandwidth profile of the generated traffic and the required QoS level in the connection setup.

The organizations that have created the standards for ATM (ITU-T and the ATM Forum) have also defined the parameters needed for a user to determine the traffic generated, and they have classified all the possible services in categories according to their QoS requirements.

7.1.1 Bandwidth Profile Characterization

The following parameters are defined by the ATM Forum. They are used to describe the bandwidth profile of the traffic that is being transmitted.

- *Peak Cell Rate* (*PCR*): The maximum rate at which cells can be transmitted in a certain connection.
- *Sustainable Cell Rate* (*SCR*): The mean transmission rate of a connection, expressed in cells per second or b/s.
- *Maximum Burst Size* (*MBS*): The maximum number of cells that a burst transmitted at a rate specified by the PCR may contain.
- *Minimum Cell Rate* (*MCR*): The minimum transmission rate guaranteed by the network.

In the ATM world, this set of parameters made up the so-called source descriptor. Not always are all of these parameters necessary to describe the source bandwidth profile. It rather depends on the service category.

7.1.2 Negotiated QoS Parameters

When setting up the connection, the originating user supplies the source traffic descriptor and also specifies the desired performance, so that the network could reserve enough resources. The parameters that define the QoS required by the user are the following:

- *Maximum Cell Transfer Delay (max CTD)*: The maximum delay that cells may experience when transmitted across an ATM network, between their generation at the source and reception at the destination.
- *Peak-to-peak Cell Delay Variation (Peak-to-Peak CDV)*: The maximum interval of variation a delay may undergo. So, if the previous parameter specifies the maximum delay, and if we subtract the value of this parameter, what we obtain is the minimum delay. Cells should experience delays that fall between these two values.
- *Cell Loss Ratio (CLR)*: The amount of cells lost in a connection as regards the total number of cells transmitted.

7.1.3 ATM Service Categories

The ATM Forum has defined five service categories. Each category is intended for applications with a specific type of traffic and with certain transmission-related QoS requirements. When a connection is established, the service category and the corresponding QoS parameters are negotiated at the same time as the path to the destination is established.

The service categories defined by the ATM Forum are:

1. *Constant Bit Rate* (CBR);
2. *Real-Time Variable Bit Rate* (rt-VBR);
3. *Non-Real-Time Variable Bit Rate* (nrt-VBR);
4. *Unspecified Bit Rate* (UBR);
5. *Available Bit Rate* (ABR);

Some of these service categories are the same as the ATM transfer capacity defined by Recommendation ITU-T I.371, although different names are used (see Table 7.1).

Table 7.1
ATM Forum service categories and ITU-T ATM transfer capacities

ATM Forum	*ITU-T*
CBR	DBR
rt-VBR	-
nrt-VBR	SBR
ABR	ABR
UBR	-
-	ABT

The CBR service is the same as the ITU-T *Deterministic Bit Rate* (DBR), and the VBR service is called *Statistical Bit Rate* (SBR), although, at the moment, only nrt-SBR is defined. The ABR service has its corresponding transfer capacity, also called ABR, but there is no equivalent for the UBR service. However, the ITU-T defines a transmission capacity called *ATM Block Transfer* (ABT), and it does not have a corresponding service category.

Each service category has a conformity definition that specifies what happens to out-of-profile traffic. The network will behave differently with excess traffic, depending on the type of service used and the status of the network. Each service has a corresponding quality-parameter priority as well. This is why, if network congestion occurs, the network may prioritize the maintenance of the quality of certain parameters before others, depending on the service used. For example, in a CBR service, the delay and delay variation values will be maintained, although the cell loss rate may increase, whereas in an nrt-VBR connection the situation is the opposite, and the lost cell rate is maintained at the expense of increasing delays.

7.1.3.1 CBR service category

The CBR service is used in connections that call for constant bandwidth. The transmitted traffic is characterized by the PCR parameter. The source may transmit cells constantly at the rate indicated by PCR, or at lower rates, or it may even remain silent.

All the cells transmitted inside the range specified by PCR have the connection-specific QoS guaranteed. This service has some particular quality requirements concerning transmission delays, delay variations and lost cells. This is why the QoS parameters max CTD, peak-to-peak CDV, and CLR must be specified.

7.1.3.2 rt-VBR service category

The rt-VBR service is aimed at those applications that have variable rates and high requirements for transmission delays and the variation of these delays. In this case, the traffic transmitted is characterized by PCR, SCR, and MBS parameters, as well as the same QoS parameters as the CBR.

7.1.3.3 nrt-VBR service category

The nrt-VBR is a service for those applications where there is no temporal relation between the two ends of the connection, and where traffic is variable in time. The same way as in the previous case, the traffic is characterized by PCR, SCR, and MBS. In this case, however, the quality of service does not have any requirements for delay, but only foe the CLR.

7.1.3.4 UBR service category

The term UBR designates the best-effort service in ATM networks. The UBR service does not provide guaranteed QoS parameters. Cells are not certified to be transmitted correctly, without any loss, nor is there any assurance of not having delays. In this case, the traffic parameter used, PCR, does not imply any QoS guarantees; the network uses it to obtain information on the maximum bandwidth used. This service is offered by legacy data networks.

7.1.3.5 ABR service category

In the ABR service, the transmission rate available for the source varies during the connection time, depending on the network status. This service guarantees a minimum transmission bandwidth, although depending on how loaded the network is, it may provide more bandwidth, up to the maximum indicated by the source. Therefore, the traffic in this service is characterized by MCR (the minimum bandwidth guaranteed) and PCR (the maximum bandwidth provided by the network, if the network is not too loaded). A minimum cell loss rate is guaranteed. However, there is no guarantee concerning delays, which means that this service is not suitable for real-time applications.

7.1.4 SLA in ATM Networks

The SLA contract defines what the client expects from the network, and it is the document the service provider will use to bill the subscriber, and manage its own network. A *Service-Level Agreement* (SLA) implies that both the user and the service provider are committed to comply with this agreement; that is, the user will adapt to it and the operator promises to provide the QoS agreed upon.

However, for congestion reasons, due to technical problems or bad network engineering, the operator might end up in a situation where it cannot provide its clients with the transmission they need. The user in his turn should not generate more traffic than is agreed to, or it could happen that the network, when transmitting this data, would be ill-treating another user and this way endangering the previous network engineering.

ATM networks have a set of mechanisms to detect when a client sends more traffic than agreed to, and what to do in such circumstances. Once this “fraud” has been detected, be it voluntary or involuntary for the client, the ATM cells violating the contract can be tagged or discarded.

To tag a cell means the following: In the header of an ATM cell there is a bit called the *Cell Loss Priority* (CLP), used to establish a first level of priority among the ATM cells transmitted. If this value is 0, this is an important cell, while if the value is 1, the cell is “not that important.” Therefore, when a client breaks the contract, the CLP of this client may be put to 1; that is to say, the cell is tagged. In this way, the network will continue carrying this cell, but with its CLP changed. If these cells arrive at a node with congestion problems, they will be among the first cells to be discarded, just like any other cell with CLP=1.

7.1.5 Resource Management

To provide the performance that users require, in addition to reserving resources during the connection setup, ATM networks implement a series of procedures to control traffic and congestion.

Resource management and traffic control are essential in making an ATM network operative and efficient. These functions try to protect the network and its users from traffic violations, and thus prevent network congestion.

ATM networks implement the following functions to manage and control the resources available:

- *Connection Admission Control* (*CAC*): This procedure is carried out when a connection is established, to determine if it can be accepted. This is the first mechanism to protect the network from congestion. The connection will either be accepted or rejected, depending on whether the network has enough resources to meet the user requirements without decreasing the performance of existing connections. This procedure looks for an acceptable path, that is, one with switching nodes and links that can support the new connection. In short, the CAC is in charge of routing the call.
- *Usage Parameter Control* (*UPC*): These are functions carried out by the network to supervise and control both the ATM connections used and user traffic. The aim, here, is to detect violations of the traffic parameter values negotiated earlier. The actions to be taken in case of violation depend on network congestion and the type of service used for transmission.
- *Network Parameter Control* (NPC): The NPC carries out the same functions as the UPC, but for links between networks (*Inter-Network Interface* or INI) instead of those between the user and the network (*User-to-Network Interface* or UNI).
- *Resource Management* (RM): Among these functions is the procedure for the network to use special cells to dynamically modify the resources assigned to a certain ATM connection.

7.1.6 The Failure of ATM

ATM was an attempt to bring QoS-aware services to a converged multimedia network, but it failed for several technical and commercial reasons:

- *ATM is more expensive than other technologies, such as Ethernet or IP.* While ATM has become a technology mainly for the backbone, Ethernet has evolved for years as a *Local Area Network* (LAN) technology, and IP has been used almost everywhere in data applications. The result is that the number of deployed ATM switches is much smaller than the number of deployed Ethernet and IP based equipment, and these technologies have become cheap and widely available.

- *Insufficient scalability.* ATM interfaces at 155 Mbit/s are common, but equipment for rates higher than this is much more uncommon and expensive.
- *Low efficiency and high complexity.* The advanced features of ATM rely on complex signalling protocols and a reduced "user data to overhead" ratio. This makes ATM a powerful but difficult and inefficient technology.

When the deployment of ATM begun, IP was already being used. In fact, the most important application for ATM is to transport IP traffic. Finally, it seems that IP and Ethernet will replace ATM. To make this possible, Ethernet has evolved to be a carrier-class technology that offers high availability, scalability and advanced *Operation, Administration and Maintenance* (OAM) functions.

7.2 QoS In IP Networks

The birth of the commercial Internet was in 1983 and since then its history has been so successful that today any communication technology must take into account the Internet and its architecture.

The Internet is no more than a mesh of communication devices (hosts) interconnected between them by means of a very versatile protocol called the *Internet Protocol* (IP). IP is in the network and inside every host as well. An important reason for the success of the Internet and the IP protocol is that hosts have implemented the IP protocol stack into their operative systems. IP has become the language of the applications. First, the hosts were all computers, but nowadays many other devices can be connected to the Internet as well, and all of them implement the IP protocol stack.

The success of the Internet is the reason why today there is so much interest in finding the way to base converged communications on IP. This implies that in the future, an evolved version of the Internet will become a global network for converged communications. The IP technology replaces ATM for this purpose.

As IP is the end-to-end technology of the new converged network, it is quite realistic to think that it would be in charge of QoS provisioning as well. This is the reason why IP is becoming QoS-aware. We are still quite far away from a global converged network, but telecom operators now have the technology needed to offer high-quality voice, video and data services over smaller IP networks.

There are two QoS architectures available for IP:

1. The *Integrated Services (IS)* architecture provides QoS to traffic flows. It relies

on allocation of resources in network elements with the help of a signaling protocol, the ReSerVation Protocol (RSVP).

2. The *Differentiated Services (DS)* architecture provides QoS to traffic classes. Packets are classified when they enter the network, and they are marked with DS codepoints. Within the network, they receive custom QoS treatment according to their codepoints only.

The IS architecture is more complex than the DS architecture, but it potentially provides better performance. One of the most important features of the IS approach is the ability to provide absolute delay limits to flows. On the other hand, the DS approach does not rely on a signaling protocol to reserve resources, and does not need to store flow status information in every router of the network. Complex operations involving classifying, marking, policing and shaping are carried out by the edge nodes, while intermediate nodes are only involved in simple forwarding operations. The IS architecture is better suited to small or medium-size networks, and the more scalable DS approach to large networks.

7.2.1 The Integrated Services Architecture

The IS architecture is an IP-network service model for both best-effort services and other, more sophisticated QoS-aware services.

The IS model assumes that the operation of all IP networks, including the Internet, must be based on:

- A common network infrastructure that relies on a single network element, the router.
- A unified protocol stack. Many protocols can co-exist above and below the network layer, but there should only be one internetwork protocol, IP.

A unified network approach makes it possible use network resources more efficiently, and network administration is easier, but this approach adds complexity to the network. Especially, to make it possible to transport both real-time and non-real-time information efficiently across an IP network, it is necessary to store some flow information in the intermediate network nodes. This is a significant modification to the original design of legacy IP networks that leave all flow processing to end systems while the core remains unaware of end-to-end connections between users. Such a modification implies upgrading or replacing legacy routers by a new generation of connection-aware routers, and this migration may be complex, especially when it comes to the Internet.

Storing the connection status is not useful, if resource reservation is not possible, and the way to reserve bandwidth in packet networks is not very evident. Flow prioritization could be used, but this fails to provide bandwidth guarantees even for

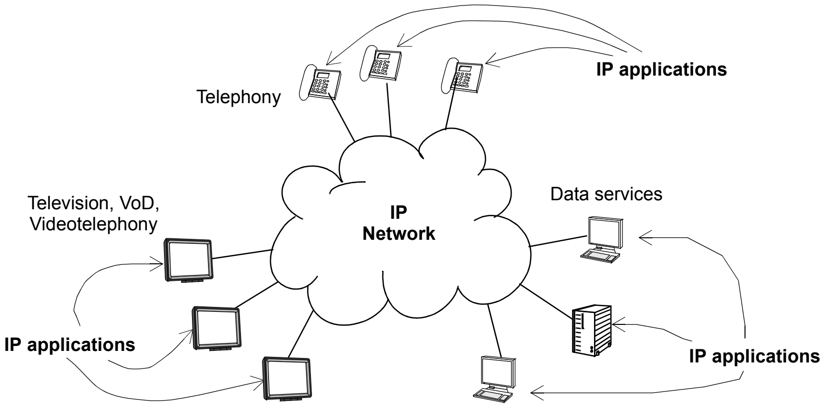


Figure 7.1 The IS architecture re-defines legacy IP networks and makes them suitable for converged voice, video and data communications. In the IS architecture, communications applications are based on IP, and IP is the transport mechanism across the network

high-priority flows. This is the reason why *advanced schedulers* are among the most important components of the IS architecture. For example, the Weighted Fair Queuing (WFQ) scheduler makes it possible to isolate flows and guarantee flexible bandwidth reservation for them.

Flows are different between them, and they must receive custom treatment in the network. Flow differentiation and custom end-to-end service types are other key components of the IS architecture. Currently, there are two end-to-end service types: controlled-load services and guaranteed services. The former provides soft QoS, the latter hard QoS.

The final component of the IS architecture is a resource-reservation protocol. This protocol must allow end systems to allocate resources for their flows and choose the right QoS parameters and service types for them.

7.2.1.1 Guaranteed Services

Guaranteed services provide assured end-to-end performance level in terms of bandwidth, delay or packet loss on conforming packets from flows with well-described bandwidth profiles.

It is clear that delay variation is small under light load conditions, due to the small probability of a packet needing to wait in a queue before being forwarded. It is also clear that under light load, a packet network will never drop packets, because the buffers of network elements will remain empty or almost empty all the time. But it needs to be demonstrated if it is possible to give absolute limits for the end-to-end performance of a network. In fact, it can be demonstrated that the WFQ scheduler can provide zero loss probability and a deterministic upper limit for the delay with traffic shaped by a token bucket. This is a theoretical result, but with very important practical implications, because it opens the door to a new class of services for packet networks.

It is possible to provide a guaranteed service by reserving B bits of buffer space and R bit/s of bandwidth in a router. If the scheduling algorithm of the routers can be approximated by GPS schedulers, the transmitter will see a dedicated line with bandwidth of R bit/s. In this case, a traffic flow conformant to a token bucket of rate r and depth b will be subject to a theoretical maximum queuing delay of b/R. A more realistic approach to the upper queuing delay limit can be achieved by adding two extra error terms:

$$Q_{max} = \frac{b}{R} + \frac{C}{R} + D \qquad \textit{eq. 7.1}$$

where C and D are two constant values. Smaller delays can be achieved, if the peak rate of the flow is limited to p. Taking into account the peak rate of the flow and the effect of packetization (packet size limited to M), the following expression for the queuing delay is obtained:

$$Q_{max} = \begin{cases} \dfrac{(b-M)(p-R)}{R(p-r)} + \dfrac{M+C}{R} + D & r \le R < p \\[2ex] \dfrac{M+C}{R} + D & r < p \le R \end{cases} \qquad \textit{eq. 7.2}$$

The assumptions made are only valid for traffic flows shaped by a token bucket. It is therefore important to perform the corresponding admission control tasks on the traffic that is subject to guaranteed limits. It may also be necessary to reshape the traffic within the network. The network administrator must decide how to deal with non-conforming packets. Often, these packets are sent without performance guarantees, but the network administrator may also decide to drop non-conforming packets.

A guaranteed service is a service for packet networks similar to a dedicated line or a switched TDM circuit. It is also less flexible and needs more resources than other services for packet networks. Guaranteed services are the best option for circuit emulation over IP. They can also be useful for provisioning of delay-sensitive applications, such as certain types of interactive video and audio.

7.2.1.2 Controlled-Load Services

Flows associated with controlled-load services are seen by the receiver as if they were best-effort flows on a lightly loaded network, even if the network is not lightly loaded.

Controlled-load services are an example of differentiated services provided on a flow basis rather than on a class basis. The effect of lightly loaded network is accomplished by isolating the flow from the rest of the traffic and devoting resources (in the intermediate routers) specifically to that flow. The amount of resources to be devoted depends on the bandwidth profile of the flow.

Although it can be expected that controlled-load services will behave better than best-effort services, there are no real guarantees for delay or packet loss performance. It is only stated that most of the packets are successfully received with a delay that is not significantly worse than the delay for those packets that do not experience congestion.

Controlled-load traffic requires shaping and policing before it enters the network. Re-shaping and re-classifying may be necessary within the network as well. What happens with non-conforming packets is a choice that has to be made by the network administrator. These packets are sometimes re-classified as best-effort packets. The alternative is to re-shape the whole flow and thus degrade conforming and non-conforming packets equally.

Controlled-load traffic offers performance-levels that fall between best-effort and guaranteed traffic. It can be used to provide interactive and non-interactive media encoded with variable-rate encoders.

7.2.2 The Reservation Protocol

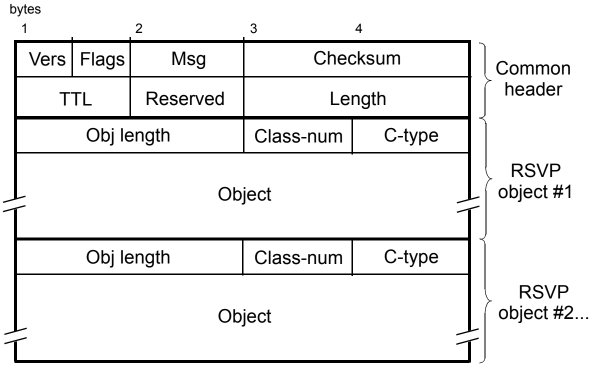
The *Reservation Protocol* (RSVP) enables end-users to reserve, maintain and release resources in IP networks. This protocol dates back to the beginning of the 1990's, and currently, it is described in various RFCs, the most important of them being the RFC 2205 released in 1997.

The RSVP has become a key component of the IS architecture. It has two main objectives:

1. Enable the provisioning of services with defined QoS needs that go beyond the best-effort service provided by legacy IP networks.
2. Deal with multicast communications beyond the basic point-to-point model of legacy IP.

The RSVP needs to interact closely with routing protocols to perform its tasks, but it is not a routing protocol. The difference is that routing protocols must look for routes to remote points in a network, and a reservation protocol must be able to manage the resources devoted to send information in the network across these routes. IP networks change dynamically as transmission conditions change. Routes change accordingly, and this may affect established reservations. The RSVP re-establishes reservations automatically when a route changes.

The RSVP does not need to be transported over UDP or TCP like most other protocols of the IP protocol stack. It can be directly encapsulated into IP datagrams with protocol number 46. Transmission over UDP is also accepted, because some end systems cannot directly map protocols into IP datagrams. All the RSVP messages have a similar format. They are made up of a common header and one or more specific fragments called *objects* that belong to a set of *object classes*. Each class has its own syntax and meaning (see Figure 7.2).



Vers: Version, currently 1 (4 bits)
Flags: Flags, currently 0 (4 bit)
Msg: Message type (8 bit)

1: Path	**5:** PathTear
2: Resv	**6:** ResvTear
3: PathErr	**7:** ResvConf
4: ResvErr	

Checksum: Error-detection word (8 bits)
TTL: Original IP TTL value (4 bit)
Length: Total message length (8 bits)
Obj length: Object length(16 bits)
Class-num: Object class identifier (8 bits)
C-type: Class type, unique within Class-num(8 bits)

Figure 7.2 RSVP message format.

A particular feature of the RSVP is that those users who make the reservation are not the generators but the receivers of the information. This is a new approach, but it is actually quite logical, because the receivers are the ones who know their own capacity limitations. Furthermore, they are the ones who experience the QoS, and in many cases they are the ones who pay for it. The receiver-initiated reservation paradigm enables heterogeneous receivers in multicast communications. Each receiver can reserve the amount of resources it is able to process. It is difficult for the transmitter of a point-to-multipoint communication to know the capacity of a large and probably dynamic group of receivers and make the reservation for them.

7.2.2.1 PATH and RESV Messages

Resource reservations in a network are made by using the RESV message. RESV messages are sent by the receiver to the transmitter, and reservations are made by intermediate routers as these messages pass through them. In certain cases, the amount of resources to reserve is known, but this is not always so. Another type of RSVP message, PATH, makes it easier to know the amount of resources to reserve. PATH messages are delivered to the receiver before sending any data. This can be done after the session negotiation of a VoIP call, for example. PATH messages specify the bandwidth profile of the transmitter, and they may also contain information on the performance of the network.

When routers find PATH messages in the network, they process them. Routers may modify certain fields of the message before forwarding it. Furthermore, intermediate routers store some information from PATH messages that generate path state entries in them. Each path state entry has a clean-up timer associated with it. When the timer expires, the path state is deleted, but if a refresh PATH message is received before the timer expires, the timer is reset. The result is that path state entries need to be periodically refreshed or the path state will be lost.

Something similar happens with reservations; they also need to be refreshed periodically. Implementing this kind of "soft-state" in the network helps to deal with the ever-changing IP network, and it is well suited to multicast communications with a large number of receivers dynamically joining and leaving communication sessions. Although receivers can free up resources by letting timers expire, they can also use the RESVTEAR message to force a situation where resources must be freed up. The transmitter can clean up the path status without the timer, by using the PATHTEAR message.

So that the RSVP would work correctly, the PATH and RESV messages must follow inverse paths between the transmitter and the receiver (see Figure 7.3). In IP networks, it is normally not possible to make sure that this condition is met. This is why PATH messages are used to store inverse routing information in intermediate routers. This information will make it possible for the RESV messages to find the correct route to the transmitter.

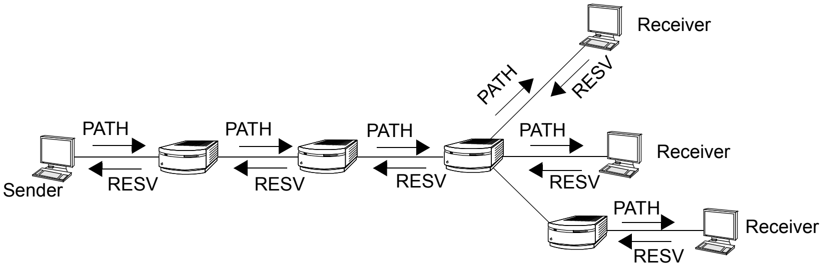


Figure 7.3 Reservations with the RSVP are carried out by the receivers by means of the RESV message, to make it easier to manage multicast sessions. Senders may use the PATH message to let the potential receivers know that there is data to be sent.

Both the PATH and RESV messages have their own error messages to inform the communicating parties on possible errors. The PATHERR message is sent upstream if the path status cannot be established in the transmission path. The RESVERR message is sent downstream when a reservation fails due to lack of resources, policy setup or for some other reason.

7.2.2.2 Making Reservations

Although reservations are made by using RESV messages, they cannot be made without a previous PATH message from the source. As explained before, PATH messages have two objectives:

1. They install inverse routing information in the path between the transmitter and the receiver.
2. They help the receiver to know which reservation to make, or force them to make certain reservations.

To do this, the PATH message uses the following objects:

- *Previous hop (Phop)*: This field is upgraded by every router with RSVP support in the transmission path. It gives the address of the last router in the path supporting the RSVP. This field needs to be updated by every router as the packet crosses the network. Thanks to this field, it is possible to know if there is end-to-end QoS support.
- *Sender template*: Intermediate routers need information to identify flows and filter them according to the reservations. This information is done by the sender template that contains the IP address and the port of the source. In IP version 6 (IPv6), the source port is replaced by the flow label.

- *Sender traffic specification (Sender Tspec)*: Contains the bandwidth profile specification of the traffic to be sent.
- Optional *Adspec (Advertising Specification)*: Contains information on the transmission path.

The RSVP specifies bandwidth profiles by means of *Tspecs* (traffic specifications). The Tspec contains the specification of the token bucket that shapes the traffic (see Figure 7.4). The token bucket specification contains the following items:

- *Token bucket rate* (r): The maximum sustainable bitrate that is allowed to pass through the shaper.
- *Token bucket depth* (b): The size of the bucket in bytes. How much traffic is allowed to pass through the shaper when the transmission rate is above the token bucket rate.
- *Peak rate* (p): Absolute upper limit of the rate (in bytes per second) that can be delivered through the shaper even if there are buckets available.
- *Minimum policed unit* (m): The minimum packet size (in bytes) needed for allocation and policing. Smaller packets are allowed, but they are considered to consume the same amount of resources as the minimum policed packet. The sender normally sets this parameter to the smallest size possible.
- *Maximum packet size* (M): The maximum size (in bytes) for a packet to be delivered. Larger packets might not be sent with QoS guarantees. The sender normally sets this parameter to the largest size possible.

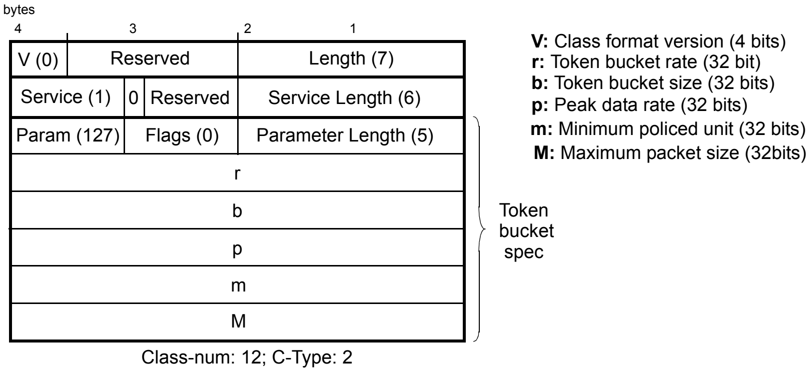

bytes
4
3
2
1
V (0)
Reserved
Length (7)
Service (1)
0
Reserved
Service Length (6)
Param (127)
Flags (0)
Parameter Length (5)
r
b
p
m
M
Token bucket spec
Class-num: 12; C-Type: 2
V: Class format version (4 bits)
r: Token bucket rate (32 bit)
b: Token bucket size (32 bits)
p: Peak data rate (32 bits)
m: Minimum policed unit (32 bits)
M: Maximum packet size (32bits)

Figure 7.4 Sender Tspec object format

While the sender's Tspec contains information about the transmitter, a PATH message may contain an Adspec object with information on the network, especially on the path between the transmitter and the receiver. The Adspec object contains a fragment with general parameters and at least one fragment with information for a guaranteed service or a controlled-load service (see Figure 7.5). If one of these service fragments is omitted, this means that the service is not available. This can be useful to force the transmitter to choose a reservation type.

The general parameters fragment of the Adspec contains the following information:

- *Global break bit:* Bit cleared by the transmitter and set when there are routers without RSVP support along the transmission path. This bit is used to tell the receiver that the Adspec may be invalid.
- *IS hop count*: The number of routers with RSVP support between the transmitter and the receiver.
- *Bandwidth*: The maximum bandwidth available for transmitting across the path. The smallest individual link bandwidth along the path.
- *Composed Maximum Transfer Unit (MTU)*: The maximum size of a deliverable packet, calculated as the minimum of all MTUs in individual links along the transmission path.
- *Minimum latency*: The sum of individual link latencies. The variable queuing delay is not taken into account, so this value represents the fixed transmission delay along the path.

The guaranteed service fragment includes the delay parameters needed to calculate the bandwidth to be allocated in the routers along the transmission path. The reserved bandwidth enables the transmitter to configure the delay (see eq. 7.2 on page 204).

- *Guaranteed service break bit*: Bit cleared by the transmitter and set when there are routers without guaranteed service support on the transmission path. This bit is used to tell the receiver that this type of service may not be available.
- *Ctot*, *Dtot*: End-to-end values of the C and D parameters of the eq. 7.2 calculated as the sum of the individual values of every intermediate router.
- *Csum*, *Dsum*: A composed value of the C and D parameters of the eq. 7.2 calculated as the sum of the individual values of all the routers since the last shaping point within the network.

The guaranteed service fragment may contain other optional items that override the values of the general parameters fragment. The controlled-load service fragment of the Adspec does not contain special parameters, but only a break bit for controlled load services, and some optional parameters that override the value of the general parameters fragment.

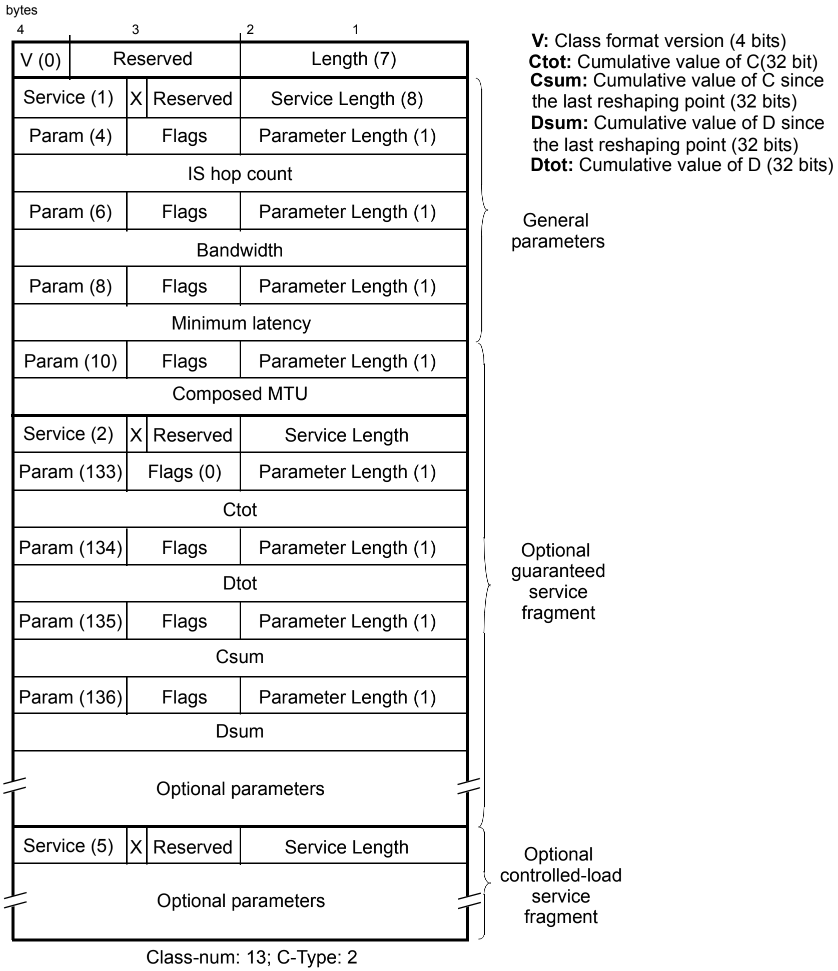


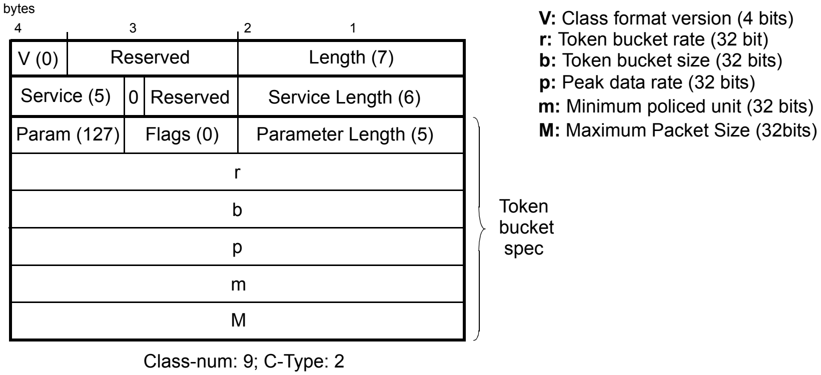
Figure 7.5 The Adspec object format

The Adspec reservation model is known as *One Pass with Advertising* (OPWA). If there is no Adspec available, the receiver can still make reservations with the help of the sender Tspec, but there is no easy way to know the properties of the transmission path in advance. This second model is called *One Pass*.

After reservation advertising with a PATH message, the receiver can allocate resources by sending RESV messages towards the origin. The RESV message contains the following objects:

- *The reservation style*: This object tells the routers how to merge reservations from different receivers in multicast communications. The value of this field depends on the application type.
- *Filter specification* (Filterspec): This object identifies the senders that are allowed to use the reservation made by the receiver. Like the sender template sent with the PATH message, this field includes IP addresses and ports or IPv6 flow labels. Sometimes any host is allowed to use the reservation; if this is the case, this object is not needed and can be omitted.
- *Flow specification (Flowspec)*: This object is used by the routers in the transmission path to calculate the amount of resources to allocate to a particular flow. The format of the Flowspec depends on the service type.
- Optional *Reservation Confirm (ResvConf)*: This object is used by the receiver to request reservation success confirmation from the last hop.

A receiver that wants to make a reservation has to generate a Flowspec according to its own preferences and sender information. To reserve resources for a controlled-load service, the receiver may simply include the sender template information in the Flowspec sent in the RESV message. In fact, the Flowspec for a controlled-load service has the same format as a the sender Tspec (see Figure 7.6). Usually, the parameter M is taken from the Adspec, if available, and not from the sender Tspec.



bytes

4	3	2	1
V (0)	Reserved	Length (7)	
Service (5)	0 Reserved	Service Length (6)	
Param (127)	Flags (0)	Parameter Length (5)	
r			
b			
p			
m			
M			

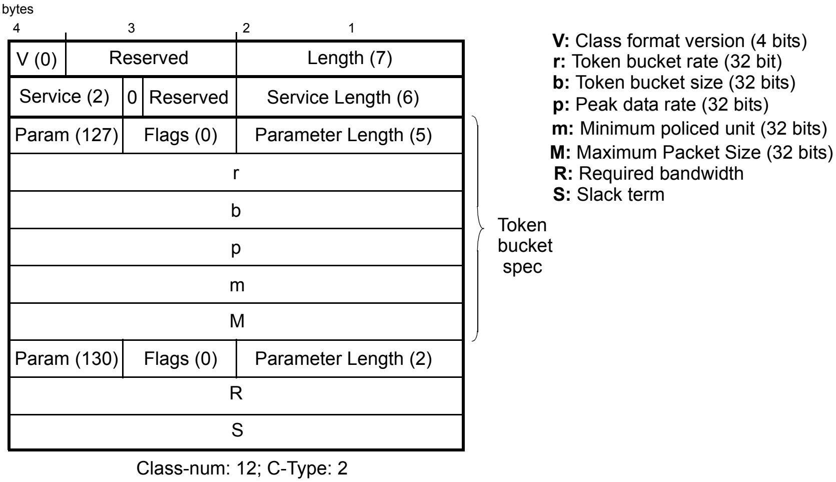
V: Class format version (4 bits)
r: Token bucket rate (32 bit)
b: Token bucket size (32 bits)
p: Peak data rate (32 bits)
m: Minimum policed unit (32 bits)
M: Maximum Packet Size (32bits)

Class-num: 9; C-Type: 2

Figure 7.6 The Flowspec object format for a controlled-load service

Making reservations of guaranteed services is slightly more complicated. The object format of a guaranteed service Flowspec has the same parameters as the controlled-load service Flowspec, but it also includes two new parameters: the required bandwidth (R) and the slack term (S) (see Figure 7.7).

1. The *token bucket rate* (r) gives information on the sustainable bit rate of the source, but to enable a service with bounded delay, it is necessary to allocate a bandwidth higher than r. This bit rate is given by R, and it is calculated with equation eq. 7.2 (see page 204) and with the parameters received in the Adspec.
2. The *slack term* (ms) makes reservations more flexible. The chances to obtain a successful reservation may be better if R is configured with a value higher than required, and the slack term set to the difference between the configured and the required value. When this is done, the routers along the transmission path may "waste" more bandwidth than needed, allowing the eventual bottleneck routers to allocate less resources than required, and still meet the delay limit.



V: Class format version (4 bits)
r: Token bucket rate (32 bit)
b: Token bucket size (32 bits)
p: Peak data rate (32 bits)
m: Minimum policed unit (32 bits)
M: Maximum Packet Size (32 bits)
R: Required bandwidth
S: Slack term

Figure 7.7 Flowspec object format for a guaranteed service.

7.2.2.3 Dealing with multiple senders and receivers

The RSVP has been designed with multicast communications in mind. This is the reason why reservations are receiver-initiated. But this is just an example, there are many RSVP-related design decisions that have been made to optimize the operation of multicast communications.

A problem that arises in multicast communications is how to deal with heterogeneous receivers. The amount of resources reserved for the same multicast session may not be the same in different interfaces of the same router. In this situation, routers need to send an RESV message to the transmitter, with the biggest possible reservation. This is to guarantee the required quality to the best receiver. The flow can be left unchanged for this receiver, but it will be shaped for those receivers that do not require high quality (see Figure 7.8). It is worth noting that not all applications may be able to tolerate severe shaping. It may be necessary to re-code the media stream before forwarding it to the receiver. This task is normally carried out by a dedicated host and not by a router.

The second problem involves applications with many senders. Some of these applications do not need to allocate separate resources for each transmitter, because it is very unlikely to have a situation where several transmitters are transmitting simultaneously. The number and identity of these transmitters may also change with time. This is why it is necessary to find out how to take the maximum profit out of the available bandwidth in this type of applications.

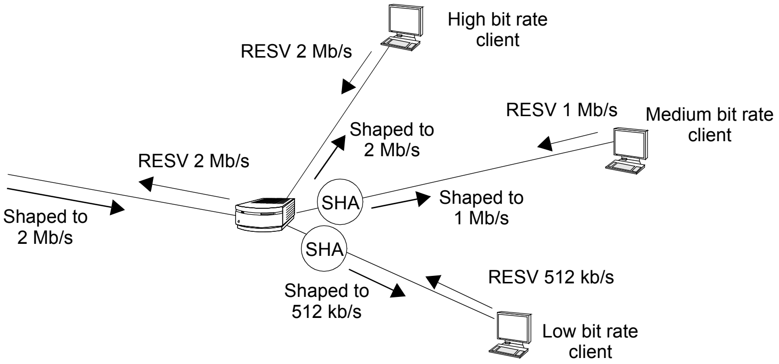


Figure 7.8 Dealing with heterogeneous receivers: The router sends the best reservation to the transmitter, and shapes the flow to meet the quality requirements of all the receivers.

The RSVP supports different reservation styles. Each of them has a different approach on the resource-sharing problem between the senders in multisender applications.

- Some share the bandwidth between different senders, and others allocate dedicated bandwidth to each sender.
- Some identify those senders that are allowed to use the reserved resources, but others are open in the sense that any sender can use the reservation.

Currently there are three different reservation styles (see Figure 7.9):

1. *Fixed Filter (FF)*: A fixed filter installs dedicated reservations for an identified sender. The reservation cannot be used by other senders, and the receiver must make one resource reservation for every sender. This style is well suited to

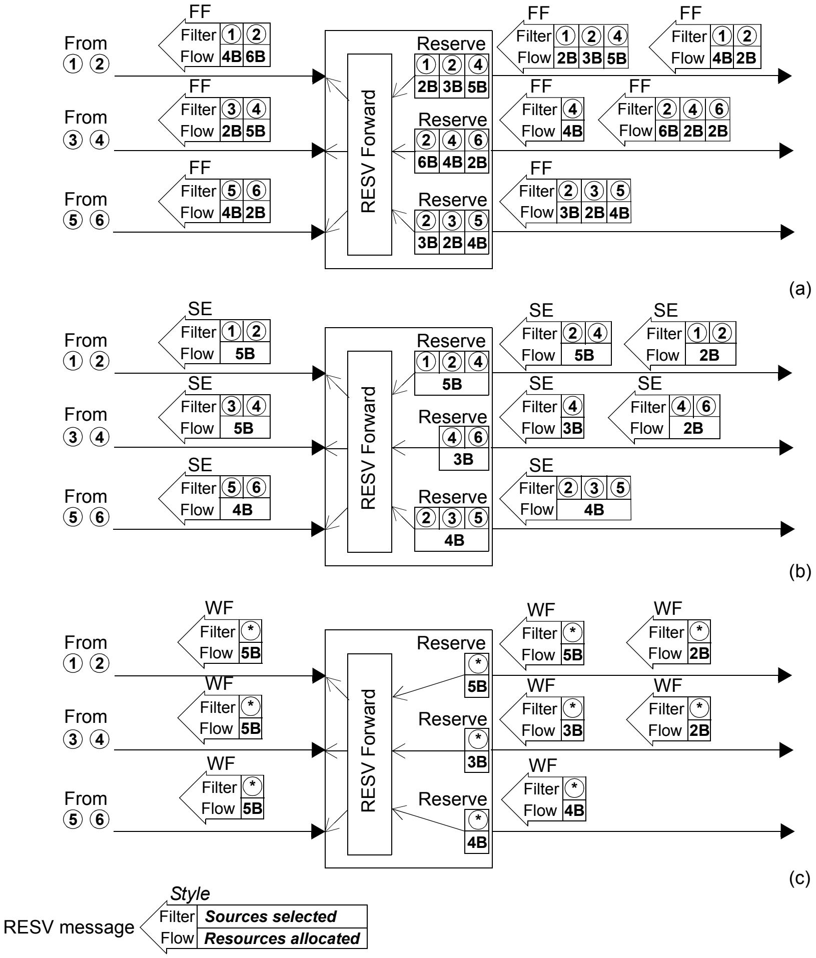


Figure 7.9 Reservation styles: (a) FF-style reservation with explicit sender selection and per-flow dedicated resources. (b) SE-style reservation with explicit sender selection and shared resources. (c) WF-style reservation with wildcard sender selection and shared resources.

video applications where many video channels are received from multiple remote sites.

2. *Shared Explicit (SE)*: In the SE reservation style reservations are shared by an identified group of senders. The Filterspec sent upstream is made up of all the senders contained in the RESV messages received from downstream. This reservation style is well suited to audio applications where the senders are well identified and stable. A good example of this reservation style are webcasts and webinars.

3. *Wildcard Filter (WF)*: The wildcard filter installs shared reservations available for an unspecified group of senders. A WF reservation is in fact a shared pipe that can be used by a dynamically changing group of transmitters. This reservation style is best suited to audio applications where the number of transmitters and their identity is either unspecified or unstable. Multiparty VoIP calls are a good example of these applications.

7.2.3 The Differentiated Services Architecture

The *Differentiated Services* (DS) QoS architecture allows for "controlled unfairness" in the use of network resources without the need for a complex signaling protocol such as the RSVP. To do this, the differentiated services architecture assumes that most of the data flows generated by different applications can be ultimately classified into a few general categories. These categories, or classes, are predefined traffic aggregates, so they are accessible without signaling. This approach attempts to be a scalable solution to the problem of providing QoS-aware services for IP networks. Scalability is achieved by migrating all complex functions to the boundary nodes, while the interior nodes remain as simple as possible. The design principles of the DS architecture make it the best option for large networks, and it should allow QoS service provisioning across large portions of the Internet.

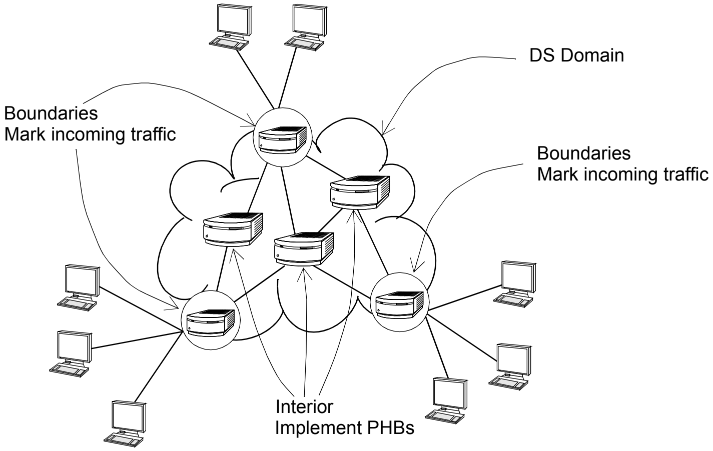


Figure 7.10 In the DS architecture, complex functions such as marking, and maybe policing and shaping as well, are transferred to the boundary nodes, while custom QoS treatment is given to packets by the interior nodes.

In the DS architecture, the boundary nodes have the following QoS-related primary functions:

- Shaping and policing the ingressing traffic to keep congestion under control.

- Marking packets according to the QoS requirements of the associated traffic flow. Marks in the DS architecture are called DS code points, and they are transported inside a special header field of the IPv4 or IPv6 datagram. The DS code points are an example of CoS labels.

Code points divide traffic aggregates into classes with different QoS requirements. Intermediate nodes of a DS domain give custom QoS treatment to packets basing on the code points carried by them, in the same way as they switch packets on the basis of their destination IP address. All QoS treatment is known as *Per-Hop Behavior* (PHB) in the DS architecture. End-to-end QoS is provided by adding individual contributions of every hop (with their individual PHBs) along the path.

Providing PHBs that are based on code points can be considered as a more developed version of the old “precedence” field of the IP datagram, or a priority-based approach. PHBs are not based on traffic prioritization, and a more sophisticated treatment is possible. Examples of treatment that goes beyond simple prioritization are: PHBs to achieve a low end-to-end delay, or PHPs to achieve controlled packet loss. Another design principle of the DS architecture is that PHB calls for buffer management of interior nodes but not route selection. This is a difference of the service-marking scheme for IPv4 datagrams based on the *Type of Service* (ToS) field.

Although traffic marking is carried out mainly by boundary nodes, interior nodes may also re-mark traffic. A typical example of this is to re-mark a part of the traffic to a lower priority class under congestion conditions.

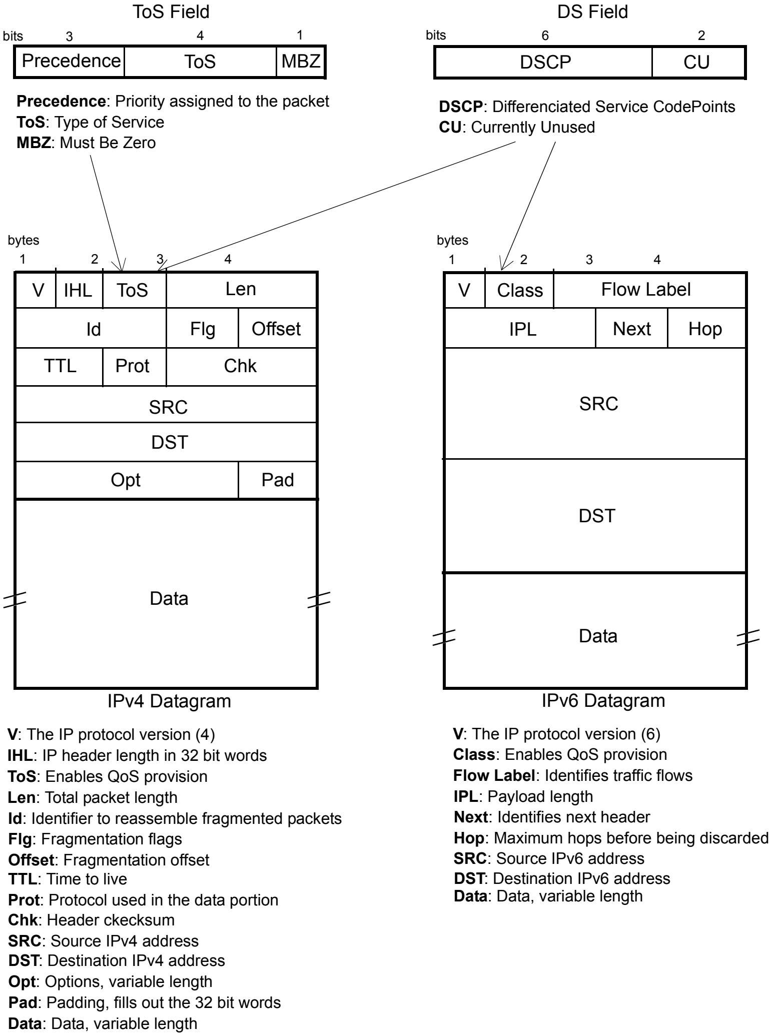
7.2.3.1 The DS Field

DS code points are transported in the 6-bit DS field of IP datagrams (see Figure 7.11). In the IPv4 datagram format, the DS field replaces the old ToS byte. The ToS byte forms a part of the IP specification since the beginning, but it has never been extensively used. The original purpose of the ToS bit was to enhance the performance of selected datagrams, to make it better than best-effort transmission QoS. To do this, a four-bit field within the ToS byte is defined, and it includes the requirements that this packet needs to meet (see Table 7.2).

Table 7.2
Meaning of ToS bits.

Binary value	*Meaning*
1xxx	Minimize delay
x1xx	Maximize throughput
xx1x	Maximize reliability
xxx1	Minimize monetary cost
0000	Normal service

In addition to the four-bit field mentioned before, there is a three-bit precedence field that makes it possible to implement simple priority rules for IP datagrams (see Table 7.3).



Precedence: Priority assigned to the packet
ToS: Type of Service
MBZ: Must Be Zero

DSCP: Differenciated Service CodePoints
CU: Currently Unused

V: The IP protocol version (4)
IHL: IP header length in 32 bit words
ToS: Enables QoS provision
Len: Total packet length
Id: Identifier to reassemble fragmented packets
Flg: Fragmentation flags
Offset: Fragmentation offset
TTL: Time to live
Prot: Protocol used in the data portion
Chk: Header ckecksum
SRC: Source IPv4 address
DST: Destination IPv4 address
Opt: Options, variable length
Pad: Padding, fills out the 32 bit words
Data: Data, variable length

V: The IP protocol version (6)
Class: Enables QoS provision
Flow Label: Identifies traffic flows
IPL: Payload length
Next: Identifies next header
Hop: Maximum hops before being discarded
SRC: Source IPv6 address
DST: Destination IPv6 address
Data: Data, variable length

Figure 7.11 IPv4 and IPv6 datagrams and the format of the ToS and DS fields, both related to QoS provisioning.

The ToS values encode some QoS requirements for the IP datagrams, but the decision on how to deal with these values is left to the network operator. For example, some operators might meet the “Minimize delay” requirement by prioritizing packets with this mark, but other operators might rather select a special route reserved for high-priority traffic.

Table 7.3
Precedence bits and their meaning

Binary value	*Meaning*
000	Routine
001	Priority
010	Intermediate
011	Flash
100	Flash override
101	Critic / ECP
110	Internetwork control
111	Network control

This is a major difference between ToS values and DS code points. While the ToS values specify the QoS requirements for the IP traffic, the DS code points request specific services from the network. Defining these services, created by means of different PHBs, is the core of the DS architecture specification.

Although there are some recommendations, most of the PHB encoding by means of DS codepoints are configurable, and they can be freely chosen by the network administrator. The only constraint for this is the backwards compatibility with the old ToS encodings.

There are some PHBs defined to be used by DS routers. The most basic of them is the *default PHB* that provides basic best-effort service and must be supported by all the routers. The recommended DS code point for the default PHB is 000000. Additionally, the *Assured Forwarding* (AF) PHB has a controlled packet loss, and the *Expedited Forwarding* (EF) PHB has a controlled delay. Other experimental PHBs are the *Less than Best Effort* (LBE) PHB for transporting low-priority background traffic, or the *Alternative Best Effort* (ABE) PHB that provides a cost-effective way to transport interactive applications by making the end-to-end delay shorter, but with higher packet loss.

7.2.3.2 Assured Forwarding PHB group

The AF PHB group provides controlled packet loss probability, as long as the traffic profile of the delivered traffic remains within the agreed SLA. There are twelve different AF PHBs, and this is why they are called a “PHB group”.

There are four AF classes. In the same router there may be packets from all four classes queuing, but each class has its own bandwidth and buffer memory resources. This means that the AF classes are isolated, and congestion in one of them does not affect the other three.

Within each AF class, there are three possible drop precedences. If there is congestion, the precedence value of the packet determines its relative importance in the AF class. Routers try to protect packets with a lower drop precedence against loss by discarding packets with higher precedence. All four AF classes together with three drop precedences per class give twelve PHBs. The code points recommended for them are listed in Table 7.3.

Table 7.4
Recommended code points for the AF PHB group

Recommended code points	*class 1*	*class 2*	*class 3*	*class 4*
Low drop precedence	001010 (AF11)	010010 (AF21)	011010 (AF31)	100010 (AF41)
Medium drop precedence	001100 (AF12)	010100 (AF22)	011100 (AF32)	100100 (AF42)
High drop precedence	001110 (AF13)	010110 (AF23)	011110 (AF33)	100110 (AF43)

The IETF specifies the AF PHB without defining any particular implementations. Network equipment vendors are free to choose the design that best suits them. Figure 7.12 shows one possible implementation for the AF PHB group. A WFQ scheduler can be used to isolate the traffic of different AF classes, but others, such as *Weighted Round Robin* (WRR) may be used as well.

The three drop precedences of every AF class are implemented by means of the *Weighted Random Early Discard* (WRED) queue management algorithm – although, again, this is not mandatory.

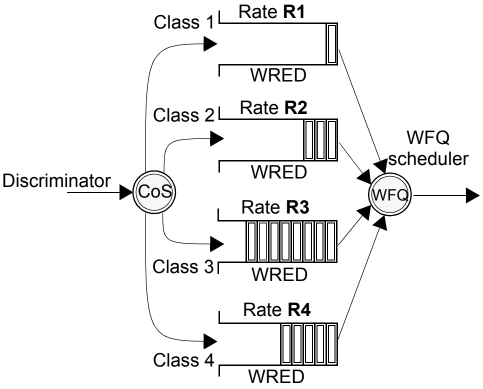


Figure 7.12 One possible implementation of the AF PHB by means of four queues with a WRED PDP and a WFQ scheduler

The AF PHB may be used to transport critical data that calls for better transmission parameters than those provided by the default PHB. Another interesting application for the AF is non-real-time video transmission. MPEG-2 video streams have medium tolerance to jitter (it must remain below 200 ms) and very low tolerance to errored, corrupted or lost packets. Specifically, the drop rate should remain below 10^{-6}. To deal with these QoS needs, video packets may be assigned to one of the AF classes. Within the video AF class, VoD traffic could be assigned to a high drop precedence like the AF12 or AF13 PHB and the IPTV traffic to a low precedence class such as AF11. The reason for this is that the VoD application needs more bandwidth than IPTV, and it is more likely to cause congestion. By assigning IPTV to a low precedence class, this application is protected against congestion caused by of VoD streams.

7.2.3.3 Expedited Forwarding PHB

The EF is intended to provide low-delay, low-jitter, low-loss services, similar to a virtual leased line. EF PHB packets receive a treatment that minimizes the queuing delay in the node. This has the following side effects:

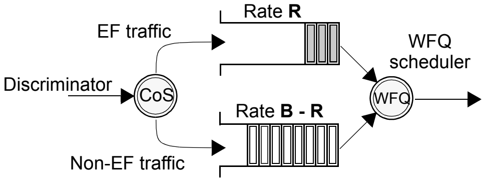
- End-to-end delay variation is also minimized, because queuing delay causes end-to-end delay and is the main cause of jitter.
- Packet loss is also minimized, because queues are not very long, and therefore the probability of losing packets is low.

The EF PHB defined in a particular node is specified by a rate, R. The rate at which EF packets should be ideally served is R or faster.

The EF PHB is similar to the controlled-load service provided by the IS architecture. Both try to emulate the behaviour of a packet-switched network under light load conditions. The difference is that while the EF PHB emulates this behaviour for an aggregate traffic class, the controlled-load service operates at flow level, offering better performance. The DS code point recommended for the EF PHB is 101110.

The EF PHB can be implemented by means of a priority scheduler. With this solution, EF packets always get privileged access to the outgoing interface. This approach could, however, lead to unacceptable performance for non-EF traffic. One possible solution is to police the EF traffic with a token bucket to limit the rate of privileged packets. Using a WFQ scheduler may also help to protect non-EF traffic from privileged traffic (see Figure 7.13).

The most important application for EF is the transport of VoIP. This application has stringent delay and jitter requirements, but it is tolerant with packet loss.



igure 7.13 A possible implementation of the EF PHB with a WFQ scheduler

Chapter 8

Broadband Access

The access network connects subscribers to the core network transport facilities, enabling end-to-end service provision. The last (or the first, depending on the point of view) portion of the access network communicates the customer premises with a local exchange. This is the part of the network known as the local loop.

The copper pair is still the leading transmission media in the local loop, followed by coaxial, fiber and wireless. *Digital Subscriber Loop* (DSL) technologies, designed for digital transmission across existing telephone copper pairs, combine cost effectiveness and acceptable performance. This fact has made DSL a very successful technology for broadband Internet access (see Figure 8.1).

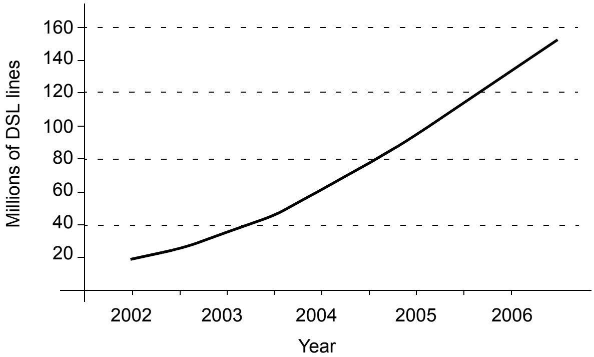


Figure 8.1 Evolution of the worldwide DSL market

Some of the latest DSLs have been designed for a specific application: Video over telephone wires. Unlike Internet access, video applications require several megabits per second for acceptable performance. One MPEG-2 encoded video stream needs about 3.5 Mbit/s. If it is accepted that 4 is likely to be the minimum number of video channels to be delivered to the customer simultaneously, the bandwidth needed just for video is about 15 Mbit/s per subscriber. If high-speed Internet access and IP telephony services are considered as well, the minimum bandwidth required for each subscriber is about 20 Mbit/s. Furthermore, it is though that in the future, *High-Definition TV* (HDTV) will require from 60 to 100 Mbit/s of bandwidth per subscriber.

The problem of DSL is that due to attenuation and crosstalk, it is impossible to achieve long range and high speed simultaneously. While it is possible to deliver a few megabits per second to customers located at several kilometers from the local exchange, delivering 100 Mbit/s is limited to a few hundred meters. This is the reason why some operators have already started to deploy new access networks based on optical fiber. Only a few of these deployments offer *Fiber To The Home* (FTTH), or (depending on where the optical link is terminated) *Fiber To The Building* (FTTB), *Fiber To The Cabinet* (FTTCab), etc.

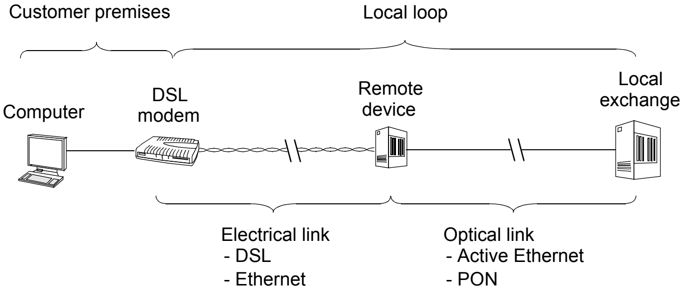


Figure 8.2 FTTx architecture for the local loop

Currently, there are many different options for FTTx. Electrical links can be built with DSL or Ethernet. The ITU-T Recommendation G.993.1 defines the *Very-high-bit-rate DSL* (VDSL), a DSL type designed for FTTCab and FTTB architectures. The VDSL technology offers downstream bit rates around 50 Mbit/s within the range of 300 meters. VDSL has been improved in the new ITU-T Recommendation G.993.2. This new technology is known as VDSL2, and it delivers symmetrical 100 Mbit/s bit rate within the range of 300 m. In FTTB architectures, the access network operator may choose to deploy Ethernet over *Unshielded Twisted Pair* (UTP) cable, if cable lengths are shorter than 100 m. The IEEE 802.3 100BASE-T and 1000BASE-T are likely to be the chosen interfaces. 100BASE-T offers 100 Mbit/s of symmetrical bit rate, and 1000BASE-T 1 Gbit/s of symmetrical bit rate.The range is limited to 100 m for both.

Active Ethernet and *Passive Optical Network* (PON) are the main options for the optical portion of the local loop (see Figure 8.3):

- Active Ethernet is made up of point-to-point fiber links between the local exchange and the customer premises. This means that large quantities of optical fiber must be used in the local loop, and this is expensive. However, the use of dedicated fiber links guarantees maximum bandwidth. To reduce the amount of fiber, an Ethernet switch can be installed close to the subscriber, and it acts as a concentrator. Between the switch and the local exchange, it is enough to install a single optical link, or maybe two for redundancy.
- PON has been proposed to avoid installing active elements, such as Ethernet concentrators, in the local loop. Active elements are replaced by simple passive optical splitters, giving as a result a point-to-multipoint topology. PON can be used to offer gigabit-level bandwidth to subscribers. This technology is considered more cost effective than active Ethernet, and at the same time it is well suited for applications like TV that can be overlapped with data on a different wavelength. The main drawback is the need for complex shared-media access mechanisms to avoid collisions between the traffic of different subscribers.

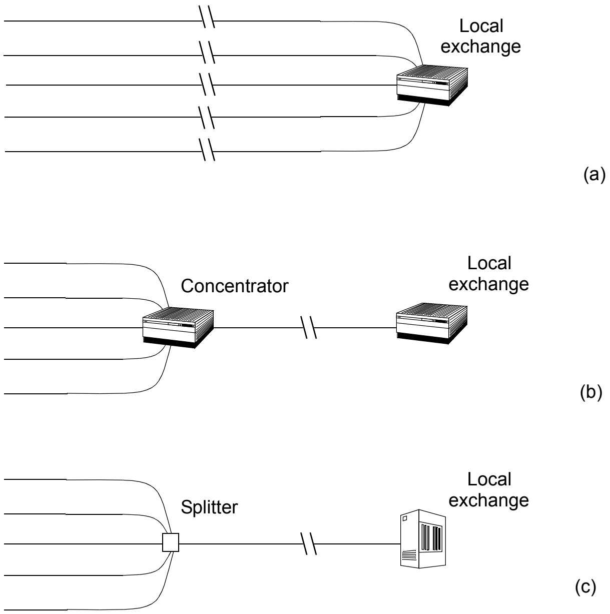


Figure 8.3 Optical fiber installation in the local loop: (a) The point-to-point topology needs a large amount of fiber. (b) With active Ethernet, less fiber is needed, because a switch can be placed close to the subscribers. (c) The PON solution replaces the switch with an inexpensive and passive optical splitter.

8.1 Broadband Services over Copper

Today, the most successful data access technology for residential customers is DSL, with more than 150 million lines installed worldwide in 2006. DSL makes it possible to reuse the existing telephone wires so that they can deliver broadband data services.

In order to work properly, DSL uses transmission bandwidth above the vocal frequency band (300 ~ 3400 Hz). There are several types of DSL (see Table 8.1), and many of them allow simultaneous transmission of analog telephone signals and data by means of *Frequency-Division Multiplexing* (FDM). DSL signals can use many Megahertz above the vocal frequency bandwidth without any noticeable disruption in the telephone service. To achieve this result, voice and data must be separated with the help of a splitter sensitive to frequency, or a similar device (see Figure 8.4).

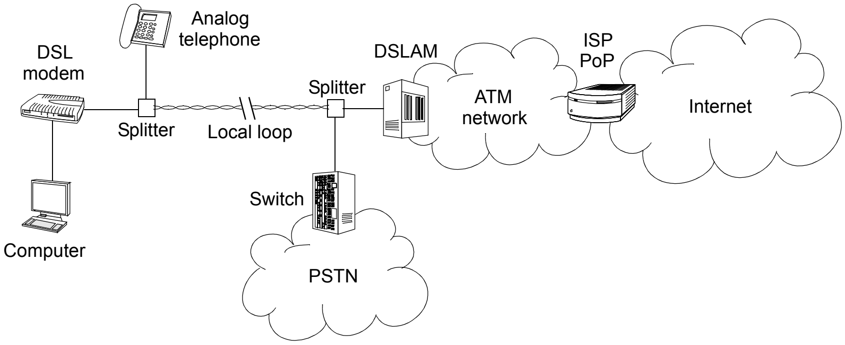


Figure 8.4 Classical access network architecture for Internet over DSL. A splitter is needed to separate the voice and data signals.

The DSL technology dates back to the late 1980s. The first DSL variant which gained wide market acceptance was the *High-bit-rate DSL* (HDSL) still used today to provide E1 (2,048 kbit/s) or T1 (1,544 kbit/s) to businesses.

Another very popular DSL technology is the *Asymmetric DSL* (ADSL), developed by the Standford University and AT&T Bell Labs. ADSL allocates more transmission bandwidth for the downstream than for the upstream. It was first designed to carry video to the homes across telephone wires. Despite the lack of success of the video application in the early 1990s, ADSL was adopted for Internet access with rates up to 8 Mbit/s downstream and 1 Mbit/s upstream. ADSL took off in the late 1990s, with massive installations in most of the developed countries. To ease and speed up ADSL roll-out, ITU-T released Recommendation G.992.2 for a special version of ADSL called G.lite. *ADSL G.lite* makes it possible for the subscriber to install the service, without an expensive visit of a specialized installer. Today, ADSL standardization and roll-out continues, and new versions such as ADSL2 and ADSL2+ enhance its performance even further.

Table 8.1
Classical broadband copper loop technologies

Name	*Description*	*Standards*	*Pairs*	*Line bit rate*	*Passband*
T1	T1 line	ANSI T1.403	Two	1544 kbit/s symmetric	0 ~1544 kHz
E1	E1 line	ITU-T G.703	Two	2048 kbit/s symmetric	0 ~ 2048 kHz
ISDN BRI	ISDN Basic Rate Interface	ITU-T G.961 ANSI T1.601	One	160 kbit/s symmetric	0 ~ 80 kHz
HDSL	High-bit-rate DSL	ITU-T G.991.1 ETSI TS 101 135 ANSI T1.TR.28	Two	1544 kbit/s symmetric	0 ~ 370 kHz
ADSL (G.dmt)	Asymmetric DSL	ITU-T G.992.1 ANSI T1.413	One	~1 Mbit/s US ~8 Mbit/s DS	25 ~ 138 kHz US 25 ~ 1104 kHz DS
ADSL (G.lite)	Splitterless ADSL	ITU-T G.992.2 ANSI T1.419	One	~1 Mbit/s US ~1.5 Mbit/s DS	25 ~138 kHz US 25 ~ 552 kHz DS
RADSL	Rate Adaptive DSL	ANSI T1.TR.59	One	~1 Mbit/s US ~8 Mbit/s DS	25 ~ 138 kHz US 25 ~ 1104 kHz DS
SDSL	Symmetric DSL	ETSI TS 101 524	One	2320 kbit/s symmetric	0 ~ 700 kHz
G.SHDSL	Single-pair high-speed DSL	ITU-T G.991.2 ANSI T1.422	One	2320 kbit/s symmetric	0 ~ 400 kHz

DSL was developed while some major changes occurred in the telecommunications market. The deregulation of the telecommunications market was intended to increase competition. New network operators, usually referred to as *Competitive Local Exchange Carriers* (CLECs), started to emerge, and they offered new services to compete with the *Incumbent Local Exchange Carriers* (ILECs) that used to be monopolistic national telecommunication companies.

Unbundling the local loop made it possible for the CLECs to use the ILECs' access facilities. The ILECs were forced to lease their copper pairs to new operators, thus enabling new operators to quickly develop their business[1]. For many CLECs, providing Internet access to homes was the killer application where they put most of

[1] Sometimes ILECs do not need to lease the copper pair completely; regulations may grant control of the DSL frequency band to the CLEC and let the ILEC control the vocal band for traditional telephony provision.

their hopes for revenue. DSL was therefore a key technology for them, and unbundling was the opportunity to quickly expand business with reduced initial investment.

8.1.1 The Limits of Copper Transmission

Despite its many advantages, DSL cannot be seen as the definitive solution for the access network. DSL has been very useful during the past 10 to 15 years, but this technology alone cannot meet the challenges of future broadband applications such as HDTV that may need up to 100 Mbit/s. DSL cannot provide long range and high transmission rate simultaneously (see Figure 8.5). Some of the most innovative DSL solutions, like the *Very-high-bit-rate DSL* (VDSL) are designed to achieve very high bit rates, but within a range limited to a couple of hundred metres.

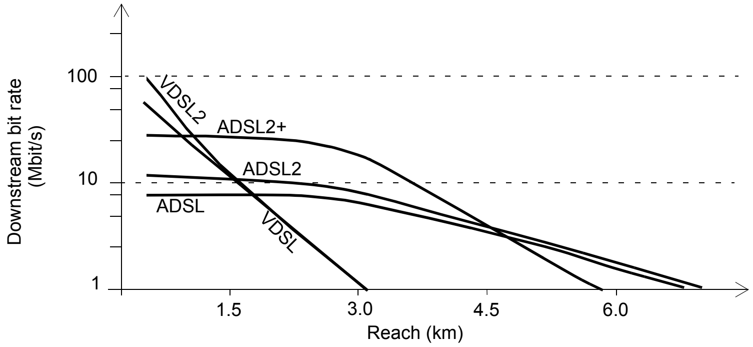


Figure 8.5 Approximate reach achieved with different DSL technologies

DSL depends on the telephone wires on which it operates, and this means that this technology has some limitations. DSL signals have to suffer many impairments in a transmission channel that was not originally designed to carry them. Two of these limitations are critical:

1. *Attenuation* is caused by progressive loss of the electrical energy of the DSL signal in the transmission line. Attenuation is higher in longer loops, and it also depends on the frequency of the signal being transmitted. The higher the frequency band used for transmission, the more attenuation the signal will suffer. Voice services may work without problems in a copper loop, but the same loop may be unable to transmit DSL signals that use higher frequency bands, due to extra attenuation in these bands.
2. *Crosstalk* is the electromagnetic coupling between transmission lines that are close to one another. In the access network, copper pairs are grouped into binders. One binder may contain dozens or even hundreds of copper pairs, and

this is why they are vulnerable to crosstalk.
There are two types of crosstalk: *Far-End Crosstalk* (FEXT) travels to the same direction as the disturbing pair, *Near-End Crosstalk* (NEXT) travels to the opposite direction (see Figure 8.6). Both FEXT and NEXT depend on the frequency. Coupling for every disturbing and victim pairs is different, and difficult to predict without testing, but generally speaking, FEXT and NEXT increase quickly when the frequency of the disturbing pairs is higher.

Attenuation is being fought against by trying to control the transmission frequency as much as possible. However, for high bit rates there is no choice, and wide portions of the copper transmission bandwidth must be used. Fighting against crosstalk is not as easy as that. Early DSL deployments, such as the first HDSLs were limited by self-NEXT; crosstalk caused by the same type of DSL at the near end of the cable. This problem was solved by introducing upstream and downstream *Frequency-Division Duplexing* (FDD). The following DSL generation was limited by FEXT and inter-system NEXT or alien NEXT.

After unbundling took place, the number of signals in the loop started to increase, and crosstalk from some local loop signals could potentially damage other operator's service. Due to this, the spectral compatibility between copper access technologies had to be studied, and new (national) regulation had to be developed to control the management of the copper loop spectrum.

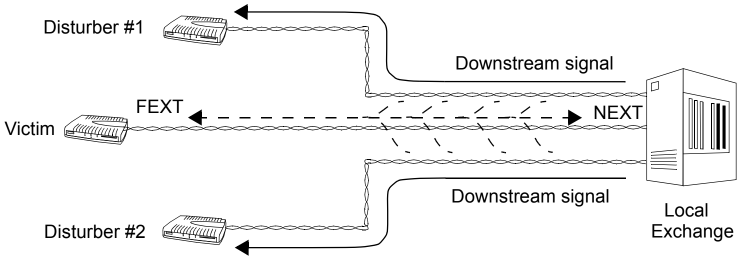


Figure 8.6 Crosstalk between copper pairs. Signals from disturbing lines are coupled to the victim line, damaging communication.

Another way to fight against crosstalk and external interferences is the use of *Discrete Multitone* (DMT) modulation. DMT is a multicarrier modulation that spreads the data to be transmitted over many orthogonal carriers instead of using a single carrier. DMT transceivers can slow down carriers affected by narrowband interferences, while unaffected carriers can continue to operate normally. The DMT technology competes with single carrier modulations, such as the simpler and more cost-effective *Carrierless Amplitude Modulation* (CAM), or the *Quadrature Amplitude Modulation* (QAM). This competition between modulation technologies led to new standards concerning different and incompatible DSLs. Currently, DMT has more acceptance than CAM and QAM. It is much more widely deployed and preferred by the standards organizations.

8.1.2 ADSL2

The second generation of ADSL (ADSL2), specified in the year 2002 in Recommendations ITU-T G.992.3 and ITU-T G.992.4, is the result of many improvements to the traditional ADSL. The signals that meet the new standard exhibit better behaviour in loops of all lengths, but the results are more impressive in shorter loops: ADSL2 delivers up to 12 Mbit/s downstream and 1 Mbit/s upstream with a range of 2 km.

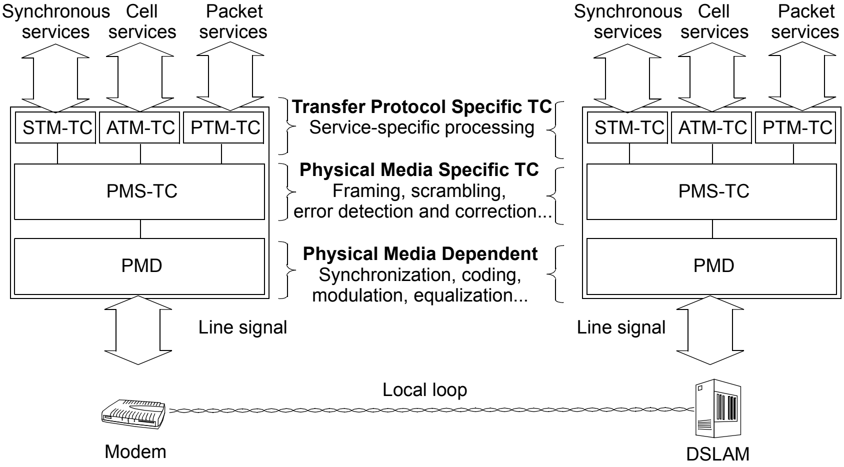


Figure 8.7 Operations involved in ADSL2 transmission and its classification

ADSL2 transceivers are quite complex and they perform many operations on the transmitted and received signals in order to ensure optimal performance (see Figure 8.7). These operations can be grouped as follows:

- *Transmission Protocol Specific / Transmission Convergence* (TPS-TC) processing: It is different depending on the kind of data being transported by DSL.
- *Physical Media Specific / Transmission Convergence* (PMS-TC) processing: Common for all client data. Includes framing, frame synchronization, error detection and correction, scrambling, and so on.

- *Physical-Media Dependent* (PMD) processing: Depends on the physical media, in this case copper pairs. Involves modulation, echo cancellation, equalization, link startup, etc.

ADSL2 has better PMD, PMS-TC and TPS-TC than ADSL. Specifically, ADSL2 includes the following important enhancements:

- *Better support of packet applications*: The new *Packet Transfer Mode* (PTM) enables direct mapping of layer 2 frames like Ethernet over ADSL2 PMS-TC frames. The PTM is added to the *Synchronous Transfer Mode* (STM) and *Asynchronous Transfer Mode* (ATM) already supported by legacy ADSL. Thanks to this new feature, packets do not need to be encapsulated in ATM cells before transmission. ATM in the local loop has always generated discussion because of the big overhead added by cells. Now it can be removed from the loop (see Figure 8.8).
- *All-digital mode*: ADSL was designed to work simultaneously with *Plain Old Telephone Service* (POTS) or *Integrated Services Digital Network* (ISDN) voice. Either POTS or ISDN is multiplexed with ADSL with the help of FDM techniques. However, many operators are considering alternative ways of delivering voice. For example, TDM-encoded voice channels can be transmitted in ADSL2 DMT tones. Another solution is to deploy VoIP solutions in the local loop and let voice share DSL bandwidth with existing data applications. Both solutions make it unnecessary to allocate a frequency band for POTS or ISDN in the local loop. ITU-T G.992.3 Annexes I and J define all digital modes for operation without POTS / ISDN. These modes are convenient for local loop with converged voice and data technology. Annex I is designed to minimize crosstalk with POTS and Annex J is spectrally compatible with ISDN.
- *Real-time rate adaptation*: Legacy ADSL systems are only able to adjust their transmission rate at initialization, but ADSL2 can seamlessly change the data rate of the connection to match the channel transmission conditions. This new feature offers a significant advantage when degradation due to crosstalk, interferences or noise varies over time.
- *Reach extended operation mode*: Annex L of ITU-T G.992.3 defines a reach-extended operation mode for ADSL2 over POTS systems. Improvements in reach may arrive up to 1 km. Annex L operation is based on specific power spectrum transmission masks with reduced bandwidth usage and increased transmitted power. For example, the downstream upper frequency is limited to 552 MHz; half of the standard upper frequency for ADSL2 over POTS systems (Annex A). New upstream and downstream masks have been carefully chosen to be spectrally compatible with existing signals.
- *Extended upstream operation mode*: Annex M of ITU-T G.992.3 defines a new operation mode for upstream bitrates up to 3 Mbit/s. To achieve this goal,

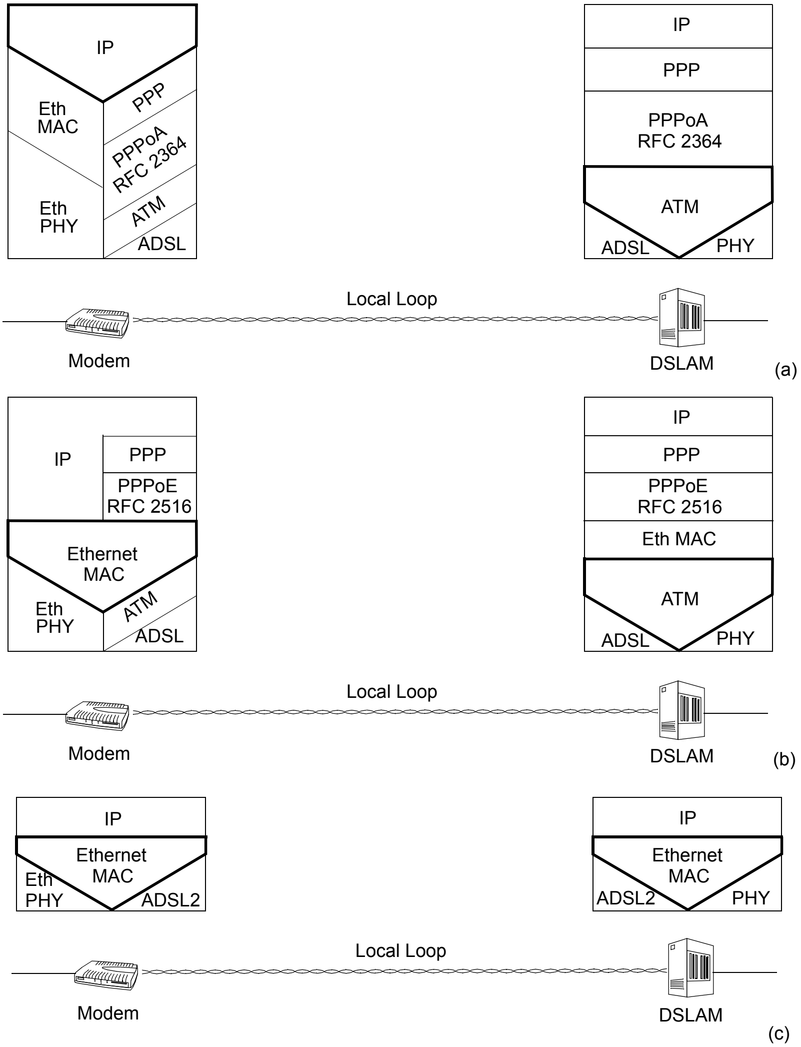


Figure 8.8 Protocol architectures in the local loop with ADSL and ADSL2: (a) Routed architecture: the user modem is a router; data is carried in ATM VCs. (b) Bridged architecture with ATM: data is encapsulated and carried over an ATM VC. (c) Bridged architecture without ATM: uses the ADSL2 PTM to map Ethernet directly in ADSL2 frames.

Annex M ADSL2 increases the number of DMT tones dedicated to upstream transmission.

- *Improved modulation and coding*: A 16-state convolutional Trellis coding is mandatory on ADSL2 DMT carriers. Furthermore, Reed-Solomon (RS) encoding of ADSL2 frames has been improved and now enables higher coding gains.

Both Trellis and RS coding add controlled redundancy to the signal, to detect and correct errors without retransmission. These features are specially helpful in longer lines, where the *Signal-to-Noise Ratio* (SNR) is low and transmission errors are likely to occur. Up to 200 m improvement in range is accomplished, thanks to modulation and coding enhancements in ADSL2.

- *Better framing*: ADSL2 framing provides more flexibility than ADSL. Frame overhead can be configured from 4 to 32 kbit/s to save bandwidth for user data when necessary. Furthermore, ADSL2 framing supports up to four different frame bearers with four latency paths.
- *Improved initialization*: The enhanced ADSL2 startup process reduces the overall handshake and training processes between transceivers in the customer premises and the local exchange to 3 seconds, while in ADSL this was about 10 seconds.
- *Power management features*: ADSL2 transceivers are equipped to make optimum usage of electrical energy. When the equipment is transmitting or receiving data, it operates in the active L0 power mode, but when transmission stops, it can go to the low power L2 mode while maintaining the ability to re-enter in L0 mode as soon as transmission starts again. ADSL2 transceivers include a sleep mode called L3 mode. The difference between the L2 and L3 operation modes is that going from L2 to L0 is very fast, and no data is lost, but going from L3 to L0 is slower.
- *Better diagnostics*: These include tools for monitoring noise, loop attenuation and SNR during and after installation. Monitoring is possible even if the channel quality is too poor to allow modem synchronization.

8.1.3 ADSL2+

ITU-T Recommendation G.992.5 was released in 2003 with a new ADSL version known as ADSL2+. This new standard is based on ADSL2, but doubling its maximum downstream rate, providing bit rates of around 24 Mbit/s in loops shorter than 1.5 km.

While the advantages of ADSL2 come from many small improvements, the performance of ADSL2+ is the result of one big design modification: The transmission frequency bandwidth is increased up to 2 208 MHz. This is twice the bandwidth of ADSL G.dmt (G.992.1) and ADSL2 G.dmt.bis (G.992.3). The ADSL2+ upstream uses the same frequency band as ADSL2, and therefore upstream is limited to 1 Mbit/s.

The benefits of increased transmission frequency are mostly noticed in shorter loops. The price to pay for this extra bandwidth is reduced range, but anyway, ADSL2+ enables quality video service provision over copper for those customers that are closer to the local exchange.

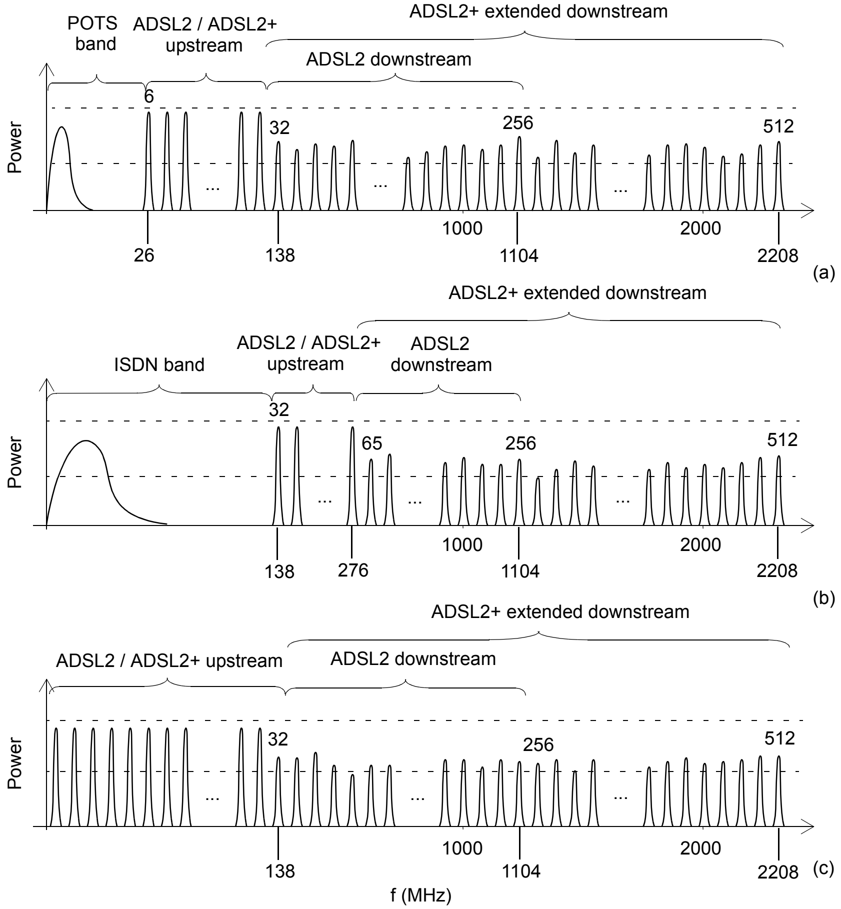


Figure 8.9 Some examples of DMT tone usage in ADSL2 / ADSL2+: (a) Tone usage for ADSL2 / ADSL2+ over POTS as specified in ITU-T G.992.3 / G.992.5 Annex A. (b) Tone usage for ADSL2 / ADSL2+ over ISDN as specified in ITU-T G.992.3 / G.992.5 Annex B. (b) All-digital mode as per ITU-T G.992.3 / G.992.5 Annex I.

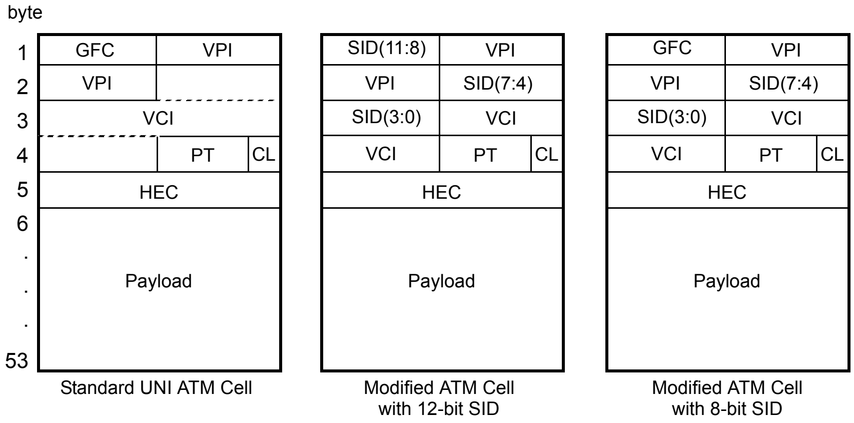
8.1.4 Bonded DSL

Bonded DSL provides a single high-speed channel by combining the capacity of two or more low-speed channels. This is done by using inverse multiplexing.

ITU-T Recommendation G.998.1, released in 2005, defines DSL bonding. This Recommendation is based on ATM, which is why it only works with ATM-TC DSL. It is also possible to bond pairs carrying Ethernet with PTM-TC DSL, thanks to the IEEE 802.3ah standard for *Ethernet in the First Mile* (EFM).

Bonded DSL can be used by service providers to deliver high bit rates to subscribers who are several kilometers away from the local exchange. So, the traditional range limitations of this technology are not an issue here. This is why those operators who wish to offer a uniform service bundle to all their potential customers may find bonded links very interesting. For example, customers closer to the local exchange may be served by a single ADSL2 / ADSL2+ loop. Those who are further away will have access to the same services, but they will be served by various ADSL / ADSL2+ bonded loops.

Bonding was enabled for the first time in the ADSL2 standard that supported *Inverse Multiplexing over ATM* (IMA). The main advantage of G.998.1 over IMA is that while IMA assumes that all the bonded links have the same nominal speed, G.998.1 allows different rates up to a ratio of 4:1 among its pairs.



Standard UNI ATM Cell

Modified ATM Cell
with 12-bit SID

Modified ATM Cell
with 8-bit SID

GFC: Generic Flow Control
VPI: Virtual Path Identifier
VCI: Virtual Channel Identifier
SID: Sequence Identifier

PT: Payload Type
CL: Cell Loss priority
HEC: Header Error Control

Figure 8.10 Cell formats with sequence numbers, as defined in G.998.1, the bonded DSL standard. There are two modified cell formats. One of them, with a 12-bit SID, can carry sequence numbers between 0 and 4095. The second format has an 8-bit SID and transports sequence numbers between 0 and 255.

Unlike IMA, G.998.1 defines a modified cell format with a new field called *Sequence Identifier* (SID). Cells are sequenced and the sequence number is inserted to the SID field before transmission. The transmitter can freely choose the link where to put every cell. The SID enables the receiver to recover the original cell stream without worrying about the specific copper pair where the cells arrive (see Figure 8.10). Bonding can be considered as a separate protocol layer placed between the physical layer and the ATM layer. It hides the diversity of transmission channels from the upper-layer protocols (see Figure 8.11). This architecture is convenient, because it is physical-layer independent, and therefore it is easy to implement in most of the current DSL technologies.

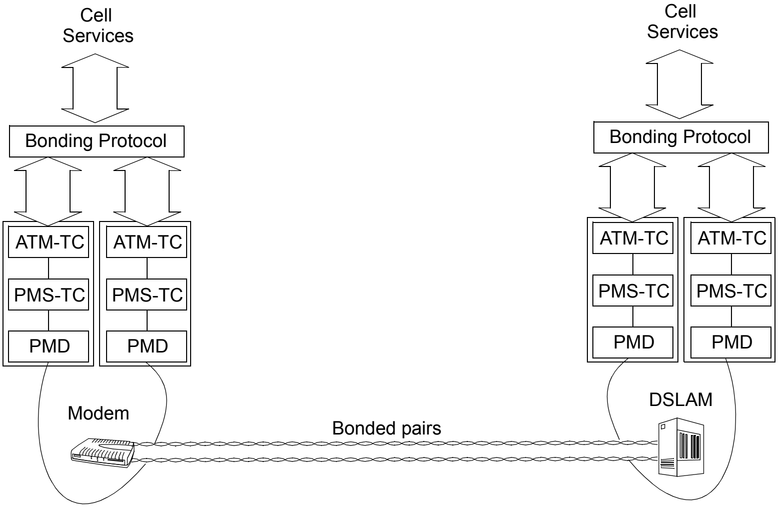


Figure 8.11 ATM cells are transported over two or more bonded copper pairs as if they were a single high-speed stream

8.1.5 VDSL

The *Very-high-bit-rate DSL* (VDSL) is a DSL technology for high speed (beyond 50 Mbit/s downstream) over short loops (usually a few hundreds of meters) by using transmission frequencies up to 12 MHz.

Early work on VDSL dates back to 1995, with simultaneous projects at ITU-T, ETSI and ANSI. Later, in 1997, the *Full-Service Access Network* (FSAN), a forum of telecom operators and vendors, developed the first VDSL specification. The market, however, did not take advantage of this work, due to a disagreement regarding the modulation technology: DMT or CAP. As a result of this, different proprietary implementations of VDSL started to appear, all of them with very limited roll-outs, mainly in Japan and South Korea.

Things did not change until 2003, when eleven major DSL equipment suppliers jointly supported DMT for VDSL, and the ITU-T could finally release Recommendation G.993.1. This Recommendation prioritized DMT – however, QAM was still included in an annex.

VDSL will never be widely deployed, because the second generation of VDSL (VDSL2) was released by the ITU-T in 2005. VDSL2 has better performance than VDSL, so all future improvements will take it as the starting point.

8.1.6 VDSL2

The second generation of VDSL, *VDSL2*, specified in Recommendation ITU-T G.993.2, is designed to increase bit rate over short and medium-length loops. VDSL2 aims to achieve the following goals:

- 100 Mbit/s symmetric service over loops shorter than 300 m
- at least 25 Mbit/s in loops shorter than 1000 m

It would not be possible to provide these rates without opening new frequency bands for digital transmission over copper. Using the band between 12 and 30 MHz increases bit rate. Frequencies between 20 and 138 kHz (optional for VDSL) are used to achieve longer reach than VDSL.

Telcos have been eager to see VDSL2, because this technology enables a new range of services over copper pairs. VDSL2 makes it possible to offer several HDTV channels simultaneously, for example. It is expected that VDSL2 will be key in enabling real competition between telcos and cable operators.

In spite of its name, VDSL2 is not just the successor of VDSL. Thanks to its performance over medium-length loops, it makes sense to migrate some ADSL2+ roll-outs to VDSL2. In other words, VDSL2 can be seen as an evolution of both VDSL and ADSL2+.

VDSL2 equipment can provide both symmetric and asymmetric services by using different configuration profiles. These profiles make it possible to share the spectrum with either POTS or ISDN, but there are also all-digital modes for converged access networks. The QAM modulation, optional for VDSL, is not used, but DMT-based implementations are used instead.

Although the ATM-TC is not fully forgotten, the PTM-TC is the main transmission mode in VDSL2. The EFM 64/65-octet encapsulation enables encoding and transmission of Ethernet frames without ATM. The VDSL2 standard improves the the EFM specification of the PTM-TC with two new features:

- Preemption is used to interrupt transmission of packets labeled as low priority when there are high priority packets waiting to be transmitted.
- Support for packets shorter than 64 bytes.

8.1.6.1 Band Plans and Profiles

Like the most popular implementations of ADSL / ADSL2 / ADSL2+, VDSL2 uses FDD for multiplexing the upstream and the downstream. Things are, however, more complicated in VDSL2, due to the broad scope of this technology. There are multiple band plans and configuration profiles to allow a wide range of services and operation modes.

These band plans and profiles are shaped by both technical and commercial factors:

- Service providers may want to offer symmetric and asymmetric services, custom QoS or other options. The configuration profiles need to accommodate all the possible services that access operators may be interested in.
- Band plans cannot be defined freely, because they are limited by the spectral compatibility between signals. Deploying new signals without considering this degrades the overall quality of both new and legacy services.
- Electronic components have their own limitations. For example, it is difficult (or at least expensive) to transmit very high power across a wide range of frequencies from a few kilohertz to dozens of megahertz. It is therefore necessary to define profiles with specific features: high power, high speed, etc.

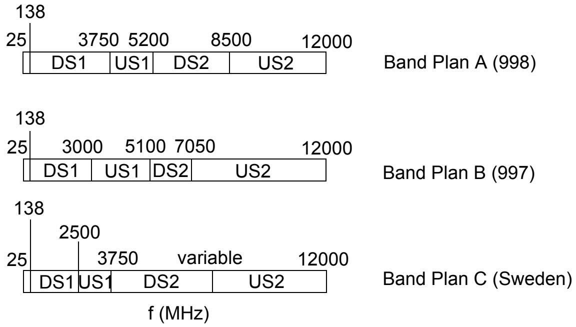


Figure 8.12 VDSL bandplans. Band Plan A (998) is better suited for asymmetric services and Band Plan B (997) is designed for symmetric services. Band Plan C is defined specifically for VDSL in Sweden.

The existing VDSL band plans served as the starting point to define band plans for VDSL2. In the year 2000, there were two main band plans for VDSL. The first one, know as Band Plan 998 was better suited for asymmetric services, and the second one, Band Plan 997, is designed for symmetric services. Both Plan 997 and 998 are based on a four-band spectrum with frequencies between 138 and 12000 kHz used alternatively for transporting upstream and downstream data. The current VDSL specification enables optional transmission in the 25-138 kHz band for upstream or downstream (see Figure 8.12).

The VDSL2 standards maintain spectral compatibility with VDSL band plans below 12 MHz. Many service providers in North America, Europe and Asia adhered to band plans based on the VDSL Plan 998, but some other European operators gave their support to spectrum usage based on Plan 997. As a result, the current version of Recommendation G.992.3 defines different deployment regions (see Figure 8.13):

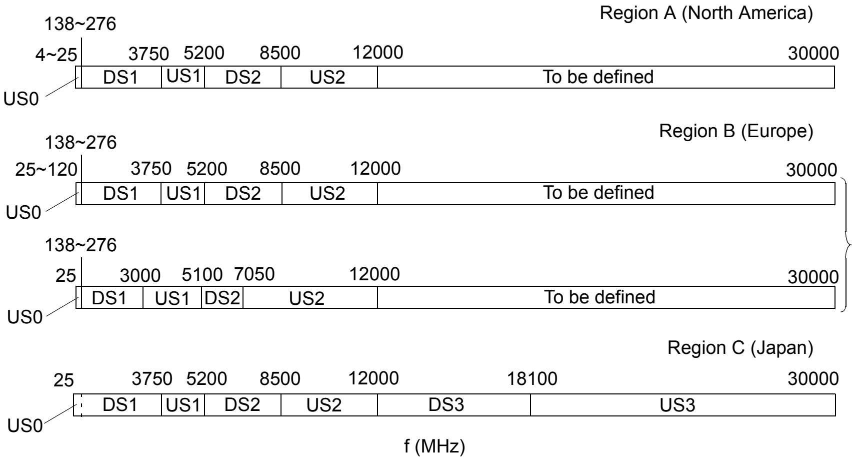


Figure 8.13 VDSL2 band plans for the three deployment regions: North America, Europe and Japan.

- In region A (North America), the VDSL2 band plan is based on Plan 998 below 12 MHz. Usage of the spectrum above 12 MHz is currently under study.
- In region B (Europe), deployments based on both Plan 997 and 998 are accepted. Usage of the spectrum above 12 MHz is to be defined.
- In region C (Japan), deployments will be based on Plan 998. Usage of the band between 12 and 30 MHz is already defined. There are two advanced VDSL2 services of 17 and 30 MHz.

It is worth noting that signals based on Plan 998 and Plan 997 are spectrally incompatible and they cannot coexist in the same plan. Working simultaneously with these signals will cause performance degradation, and as a result, services will stop working as expected. This is more likely to occur in Europe, where two incompatible band plans are accepted.

Another issue covered in the definition of the different band plans is the usage of the low frequency band between 20 and 138 Hz (US0 band). This band is currently regulated as optional for VDSL, but it is required in long-reach VDSL2 configuration profiles.

ITU-T Recommendation G.993.2 defines eight different profiles or operation modes for VDSL2 (see Table 8.2). These profiles can be considered as setups of VDSL2 systems, as they contain configurations for the transceivers. The spectrum bands to be used, the maximum and minimum transmitted power and other parameters are given for every profile to achieve specific purposes while taking into account the limitations of the physical components. The spectrum usage of the profiles is different, depending on whether the service is deployed according to Plan 997 or 998.

Table 8.2
VDSL2 Profiles

Name	*DS Power (dBm)*	*US Power (dBm)*	*Max DS Freq (MHz)*	*Max. US. Freq (MHz)*	*US0 usage*	*Carrier Spacing (kHz)*
8a	+17.5	+14.5	8.5 (plan 998) 7.05 (plan 997)	5.2 (plan 998) 8.8 (plan 997)	Required	4.3125
8b	+20.5	+14.5	8.5 (plan 998) 7.05 (plan 997)	5.2 (plan 998) 8.8 (plan 997)	Required	4.3125
8c	+11.5	+14.5	8.5 (plan 998) 7.05 (plan 997)	5.2 (plan 998) 5.1 (plan 997)	Required	4.3125
8d	+14.5	+14.5	8.5 (plan 998) 7.05 (plan 997)	5.2 (plan 998) 8.8 (plan 997)	Required	4.3125
12a	+14.5	+14.5	8.5 (plan 998) 7.05 (plan 997)	12	Required	4.3125
12b	+14.5	+14.5	8.5 (plan 998) 7.05 (plan 997)	12	Not Required	4.3125
17a	+14.5	+14.5	17.7 (Japan)	12 (Japan)	Not Required	4.3125
30a	+14.5	+14.5	18.1 (Japan)	30 (Japan)	Not Required	8.625

There are four 8-MHz profiles (8a, 8b, 8c and 8d) that do not use transmission bands above 8 MHz. Transmission over the US0 band is required for these profiles, however. Profiles 8a and 8b have increased downstream transmission power to improve performance when operating in strong crosstalk environments.

8 MHz are well suited to be installed in the central office without the need to use optical fiber in the local loop, with performance compared to ADSL / ADSL2 / ADSL2+. The two 12-MHz profiles (12a and 12b) may be used to provide broadband services either from the local exchange or in *Fiber To The Cabinet* (FTTCab) rollouts. Finally, the 17-MHz (17a) and the 30-MHz (30a) profiles are specifically designed for very high rates in deep fiber roll-outs. The 17-MHz profile applies to FTTCab, and the 30 MHz profile to FTTB. Performance in terms of reach and speed are diverse, ranging from ADSL2+-alike performance to the 100 Mbit/s symmetric service over very short loops (see Figure 8.14).

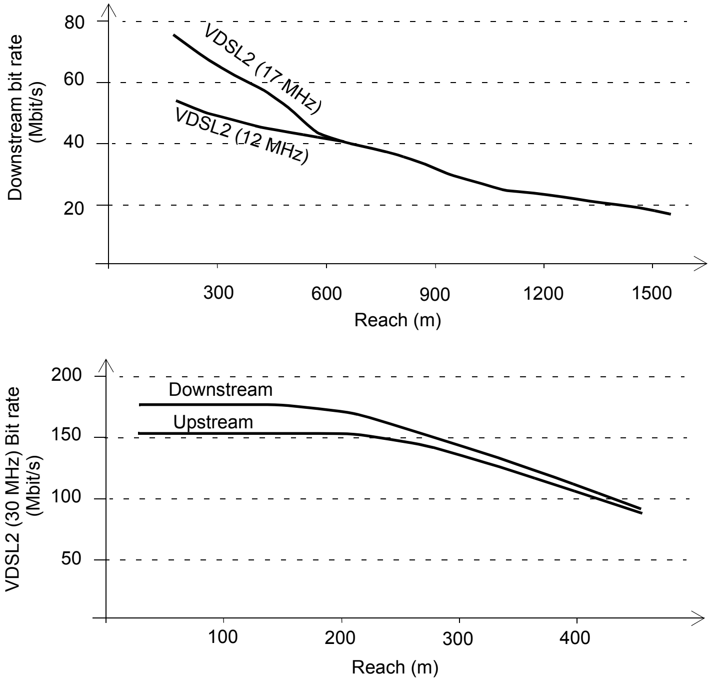


Figure 8.14 Reaches achieved with different VDSL2 profiles. Measurements are collected on a 0.4-mm wire in presence of 24 VDSL2 disturbers.

8.1.6.2 VDSL2 in Deep Fiber Rollouts

The range of high-speed VDSL2 profiles is shorter than the average distance between the local exchange and the customer premises. This means that VDSL2 cannot achieve its maximum performance without moving the *DSL Access Multiplexer* (DSLAM) out of the exchange.

New DSLAMs, known as remote DSLAMs, can be installed in street cabinets, in the basements of buildings, or in other locations close to the subscribers. By making the distance between the DSLAM and the subscriber shorter, it is possible to increase the bit rate and get the maximum performance from VDSL2. Deploying remote DSLAMs means that optical fiber links must be installed between DSLAMs and the local exchange. Although any optical technology is valid for these links, PON and optical point-to-point Ethernet are the most popular solutions today.

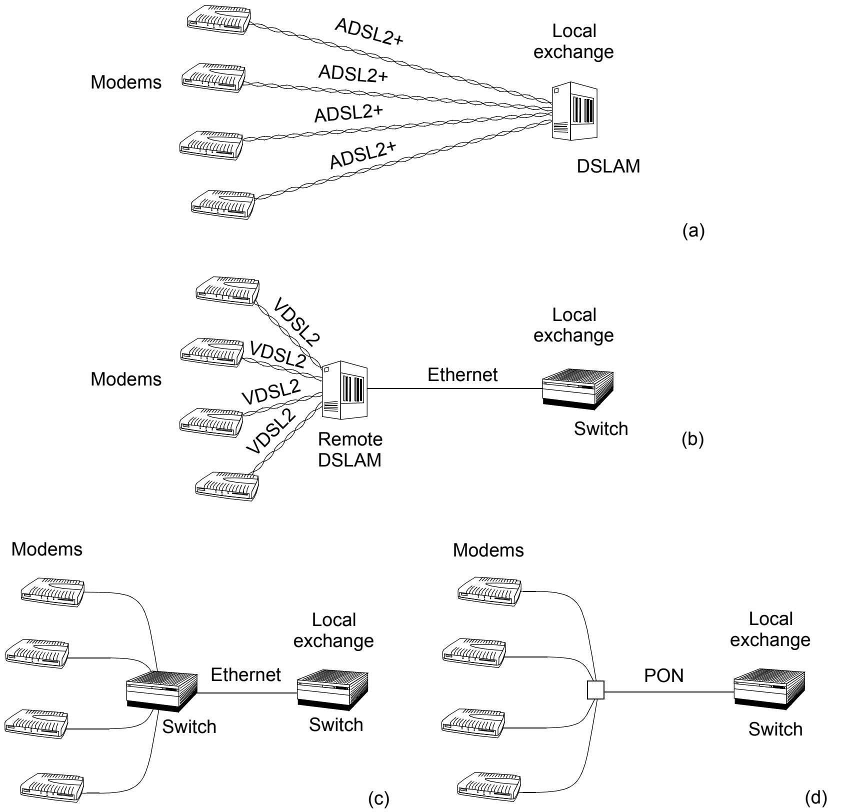


Figure 8.15 Smooth transition to an FTTH architecture: (a) The signal is delivered from the local exchange by means of copper pairs and ADSL2+. (b) Remote DSLAMs are used, and now the signal can be delivered at a much higher rate with VDSL2 links. (c) The remote DSLAM is replaced by a switch with optical ports, and the local loop becomes an Ethernet network. (d) The remote DSLAM is replaced by an optical splitter, and subscribers are serviced with PON links.

External plant deployment of DSLAM involves several new challenges that need to be solved:

- The equipment must be placed in a suitable place, far from electromagnetic interference. The place must also be secure, to prevent any access by unauthorized people.
- DSLAMs must be prepared for difficult weather conditions. For example, while in the local exchange typical temperature conditions range from -5 ºC to +40 ºC, remote DSLAMs must be prepared for temperatures between -40 ºC and +65 ºC.
- Suitable electrical power sources must be found. Sometimes, remote feeding can be used. In FTTB roll-outs, access to local power sources is an alternative. Using redundant power sources is recommended.

VDSL2 rollouts with remote DSLAMs are considered by many operators as the first step in the migration path to an access network entirely based on optical communications (see Figure 8.15). Installing optical fiber is expensive, and the cost per subscriber of FTTH deployments is often difficult to justify. Other deep fiber roll-out strategies based on VDSL2 over existing copper wires can be used to reduce the investment per subscriber and simultaneously enable some bandwidth-demanding services such as TV or network gaming.[2]

Existence of VDSL2 transmitters in cabinets may potentially cause disruption in the services transported across local exchange pairs. Crosstalk injected in pairs transporting attenuated signals will most likely damage them, even if the victim pairs are spectrally compatible like ADSL2+ and VDSL2 (see Figure 8.16).

To deal with this problem, the power spectrum of VDSL2 signals is shaped by a filter that reduces the power to a convenient level in the frequency band between 138 kHz and 2208 MHz, the band shared by ADSL2+ and VDSL. The amount of VDSL2 attenuation can be programmed: it will be more intense when the remote DSLAM is placed further from the local exchange, and thus victim signals are weaker.

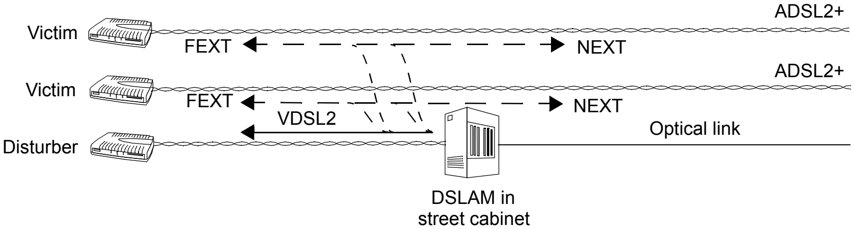


Figure 8.16 Crosstalk injected into ADSL2+ pairs by cabinet signals like VDSL2 is likely to cause damage in victim pairs, even if the signals are spectrally compatible.

[2] Market development may justify massive installation of optical fiber to the customer premises.

8.2 The Passive Optical Network

The *Passive Optical Network* (PON) is an optical technology for the access network, based only on passive elements such as splitters. In a PON, the transmission medium is shared, and traffic from different stations is multiplexed. Optical transmission increases transmission bandwidth and range dramatically when compared to some copper pair technologies such as DSL. Furthermore, due to the use of simple and inexpensive transmission elements and shared medium, a PON is a cost-effective solution for the optical access network.

The logical deployment alternative enabling optical communications in the local loop is to replace the copper links by optical fiber links, but this requires a lot of fiber. Installing Ethernet switches acting as traffic concentrators near the customer premises requires less fiber, but massive installation of Ethernet switches has the same inconveniences as remote DSLAMs: suitable placement and power supply must be provided. This is one of the reasons why PON, based only on passive elements that do not need feeding, is a very attractive solution.

Preliminary works on the PON technology date back to the late 1980s, but the first important achievement regarding its standardization did not arrive until 1995. This year, the *Full-Service Access Network* (FSAN) was formed and presented a system specification for *ATM PON* (APON). Later, in 1997, the ITU-T released Recommendation G.983.1 based on the FSAN specification. The APON is known today as *Broadband PON* (BPON) to emphasize that although ATM-based, any broadband service can be provided with this technology.

Since the release of Recommendations G.994.x for *Gigabit PON* (GPON) in 2003, APON / BPON is considered a legacy technology. GPON has been specified with the help of the FSAN, and it provides multigigabit bandwidths at lower costs than BPON, while achieving more efficiency transporting packetized data with the new lightweight *GPON Encapsulation Mode* (GEM). The GEM is based on a concept similar to the *Generic Framing Procedure* (GFP), a successful encapsulation for mapping packets in SDH networks.

Table 8.3
PON Technology Comparison

	APON / BPON	*GPON*	*EPON*
Downstream rates (Mbit/s)	155, 622	1244, 2488	1000
Upstream rates (Mbit/s)	155, 622	155, 622, 1244, 2488	1000
Range (km)	20	20	20
Encapsulation	ATM	GEM / ATM	Ethernet

An alternative approach is the *Ethernet PON* (EPON), released in 2004 as a part of the IEEE 802.3ah standard for Ethernet in access networks. The main innovation of EPON is that it encapsulates data in Ethernet MAC frames for transmission. Today, EPON has become a strong competitor for GPON, and there are supporters and deployments for both technologies (see Table 8.3).

8.2.1 Basic Operation

The physical properties of passive optical splitters make the distribution of optical signals with PON different from other technologies with a shared access to the transmission medium. Ports in optical splitters do not all have the same properties, and thus the network elements connected to them are different:

- The *Optical Line Termination* (OLT) is connected to the uplink port of the optical splitter. Any signal transmitted from the OLT is broadcast to all the other ports of the splitter.
- The *Optical Network Unit* (ONU) is connected to the ordinary ports of the optical splitter. When signals transmitted from the ONT arrive to the splitter, they are retransmitted to the uplink towards the OLT, but not to other ordinary ports where other ONUs could be connected. This makes direct communication between ONUs impossible.

The OLT constitutes the network side of the PON, and it usually resides in the local exchange. The ONUs are the user side. They can be placed in the customer premises in FTTH roll-outs, but they can also be deployed in cabinets, basements of buildings or other locations close to the subscribers. In cases where the ONU is not directly available to the subscribers, the signal is delivered to them by means of other technologies such as DSL or Ethernet. The ONU in FTTH is sometimes referred to as *Optical Network Termination* (ONT).

In the PON technology, the upstream and downstream signals can be multiplexed by physically separating the signals into different optical fibers, or by using - *Wavelength-Division Multiplexing* (WDM) mechanisms, i. e. by transmitting the upstream and downstream signals over the same fibre but at different wavelengths.

Signals from two or more ONUs transmitting simultaneously will collide in the uplink, and the OLT will be unable to separate them, unless a bandwidth-sharing mechanism is implemented. WDM appears to be the most natural way to share the transmission media for PON, but it would require either installing tunable lasers in the ONUs or having many different classes of ONUs for transmitting at different wavelengths. The high cost of the first solution and the complexity of the second one make WDM-PON unfeasible today, but attractive in the future.

Transmission in current PONs is based on TDM rather than on WDM. TDM allows for a single downstream wavelength, but it relies on complex shared-media access algorithms. Contention based on the *Carrier Sense Media Access / Collision Detection* (CSMA/CD) algorithm was successfully used with broadcast Ethernet, but it cannot be used with PON for the following reasons:

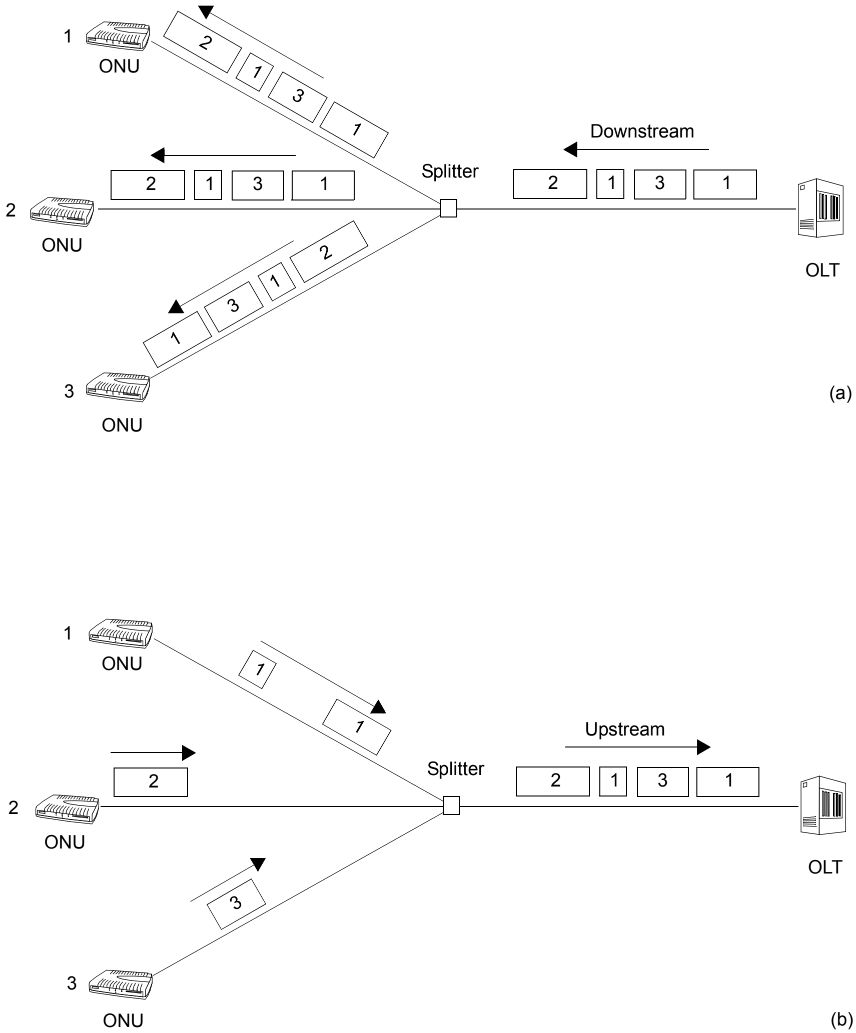


Figure 8.17 Transmission medium sharing a PON: (a) The downstream signal is broadcast to all the ONUs. (b) The upstream signal is point-to point. The section between the splitter and the OLT is shared between all the ONUs.

- ONUs cannot receive signals from other ONUs, and therefore they are unable to detect collisions. The OLT is the only station able to detect collisions between stations.
- The performance of CSMA/CD is closely related to the network size. CSMA/CD can be used in LAN environments with size limited to hundreds of meters, but it would not work in PONs of dozens of kilometers.

There are bandwidth sharing mechanisms specifically designed for PON. These algorithms take into account that only communications from an ONU to an OLT, but not between ONUs are possible, and therefore they assign to the OLT controller functions. The OLT decides which ONUs are allowed to transmit, when they are allowed to do it, and how much data are they allowed to transmit upstream. The decisions made by the OLT must avoid collision even in the case of propagation delays, and at the same time they must grant fair bandwidth sharing and high network usage. All the transceivers in the PON must be synchronized to a common time reference in order to work properly. The OLT is the network element that is usually in charge of distributing synchronization.

The downstream of a PON is dedicated, and thus no bandwidth sharing mechanisms need to be implemented. However, the downstream link is a broadcast channel, and information transmitted by the OLT is received by all ONUs even if this information is not addressed to all of them. This has some privacy implications and makes it necessary to encrypt private downstream data.

8.2.2 Advantages

PON offers increased bandwidth and range when compared to DSL. It is also more cost-effective and easier to maintain than active Ethernet. It also has several other advantages, namely:

- PONs are highly transparent, as the optical distribution network only contains layer-1 devices. Virtually any type of service can be built over PONs, either packet, TDM or wavelength-based, or even analog. Transparency eases migration to new technologies without the need to replace network elements. For example, migration to WDM PON would require replacing end equipment, but not the optical distribution network.
- The PON point-to-multipoint architecture in the downstream makes it easy to offer broadcast services such as TV. Broadcast services can be provided in a dedicated wavelength separated from unicast and multicast data services.
- There are many topologies compatible with the PON technology beyond the basic star topology. Various 1:N passive splitters can be chained, allowing for a tree topology. Using 1:2 tap couplers enables bus and ring topologies. Further-

more, basic topologies can be easily extended to redundant topologies offering resiliency when facing service shortages (see Figure 8.18).

On the other hand, using PONs has some inconveniences as well. The most important drawbacks are reduced range and bandwidth when compared to active Ethernet, due to the attenuation introduced by the splitters and the effect of sharing resources.

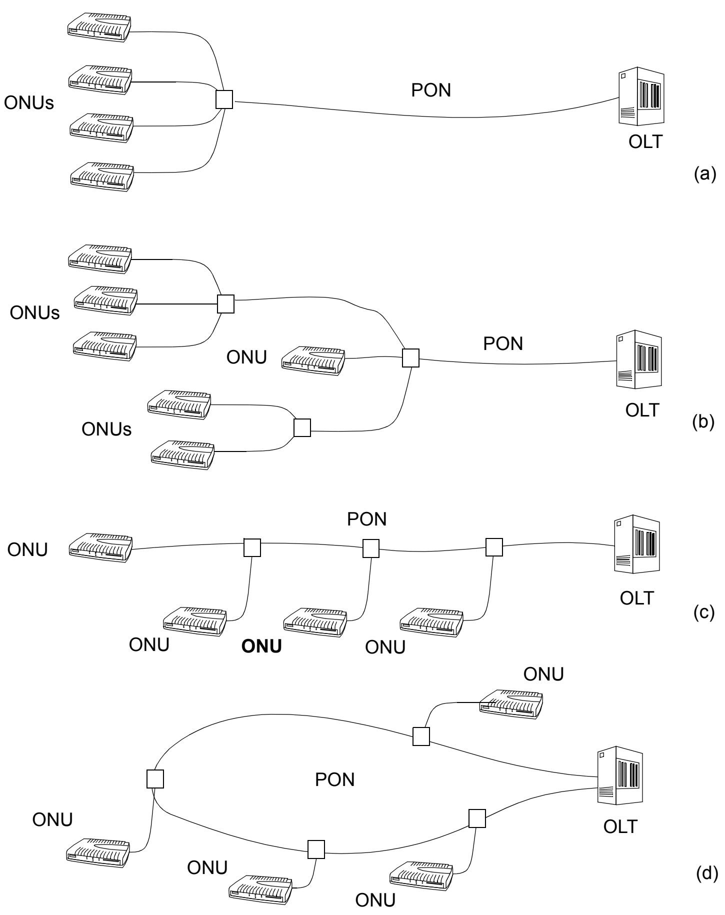


Figure 8.18 Different PON topologies: (a) star (b) tree (c) bus (d) ring

8.2.3 Broadband PON

Broadband PON or BPON was the first version of a PON to be included in an international standard in the mid-1990s. By that time it was the logical decision to choose an ATM encapsulation for BPON (or APON, as it was known before). BPON was specified by the FSAN, a group of telecommunications companies and telecom equipment vendors. For them, it was natural to choose a technology specifically designed for carrier applications. From the unexpected success of Ethernet and IP arose the need to modify the original design to provide better support of these technologies.

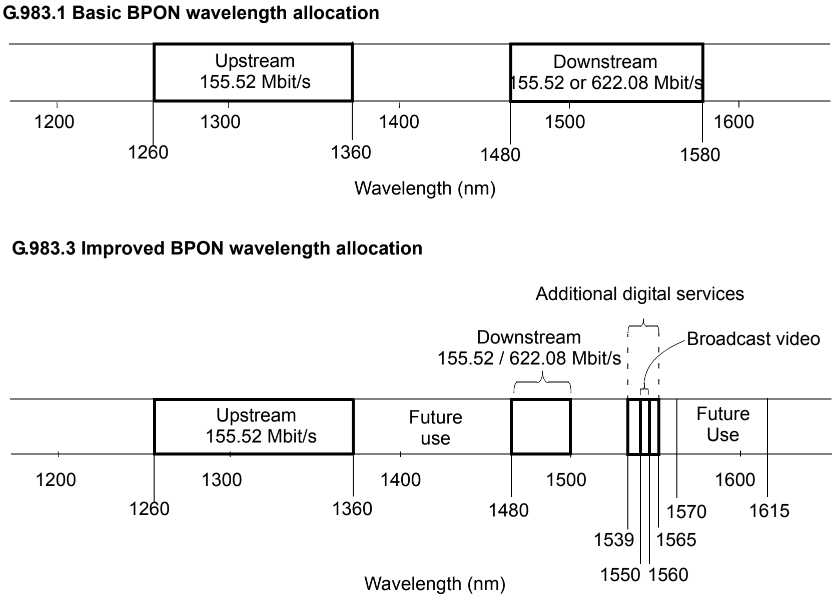
Current version of BPON delivers upstream rates of up to 622.08 Mbit/s, and downstream rates of up to 1244.16 Mbit/s with a range of up to 20 km. However, the most common versions are the symmetrical 155.52 Mbit/s and the 155.52 / 622.08 Mbit/s. These are aggregated bit rates that must be shared by all users in the BPON. A single BPON can serve up to 32 ONUs.

Recommendation G.983.1 allows for spatial and wavelength duplexing of both upstream and downstream. In the first case, signals are allocated in the 1260-1360 nm wavelength window, whereas in the second case, the upstream is kept in the same window but the downstream is moved to the 1480-1580 nm window. Recommendation G.983.3 improves this bandwidth allocation by defining two new bands for additional digital services and video broadcasting (see Figure 8.19)

The upstream and downstream in the BPON are continuous streams of cells. The downstream is broadcast and it contains cells for all users. ONUs must extract from the stream the cells addressed to them and discard others. BPON provides downstream privacy by scrambling the contents of every OLT / ONU downstream point-to-point connection with an algorithm that uses a 3-byte key known as churn key. The churn key is provided by the ONUs and updated periodically.

The upstream is shared by all the ONUs using TDM mechanisms. Permission for ONUs to transmit must be granted by the OLT. Grants are inserted downstream in special cells called *Physical Layer Operation, Administration and Maintenance* (PLOAM) cells. One PLOAM cell is inserted downstream every 28 ATM cells. The OLT grants transmissions to avoid collisions between ONUs and guarantees optimum usage of the transmission channel.

ONUs are subject to transmission delays. The delay may be different for each ONU, depending on the network topology, and the OLT must take this into account to avoid collisions. To do that, the OLT measures the distance to the ONUs, and then requests the ONUs to insert the appropriate delay so that all equivalent ONU / OLT distances are 20 km. This process is called ranging.



G.983.1 Basic BPON wavelength allocation

G.983.3 Improved BPON wavelength allocation

Figure 8.19 Optical bandwidth usage by BPON with wavelength based duplexing of the upstream and the downstream.

8.2.4 Gigabit PON

Gigabit PON or GPON is an ITU-T standard for PON that enables upstream and downstream bitrates above 1 Gbit/s. More specifically, GPON provides 1244.16 / 1244.16 Mbit/s, 1244.16 / 2488.32 Mbit/s and 2488.32 / 2488.32 Mbit/s transmission interfaces. Interfaces at 155.52 and 622.08 Mbit/s are also possible for the upstream when higher bitrates are not needed. GPON cannot interoperate with BPON, even if the line rates are the same for the transceivers.

BPON and GPON have similar optical specifications. Both single and double fiber transmission is possible. In the first case, the upstream and downstream are allocated in the optical window between 1260 and 1360 nm. In the second case, the downstream is moved to the window between 1480 and 1500 nm. Optical transmission features provide a maximum range of 20 km between the OLT and the ONU.

The main improvement of GPON over the legacy BPON is the new *GPON Encapsulation Mode* (GEM) that enables transport of TDM and packets like Ethernet without ATM (see Figure 8.21). The GEM is very efficient: It adds just five over-

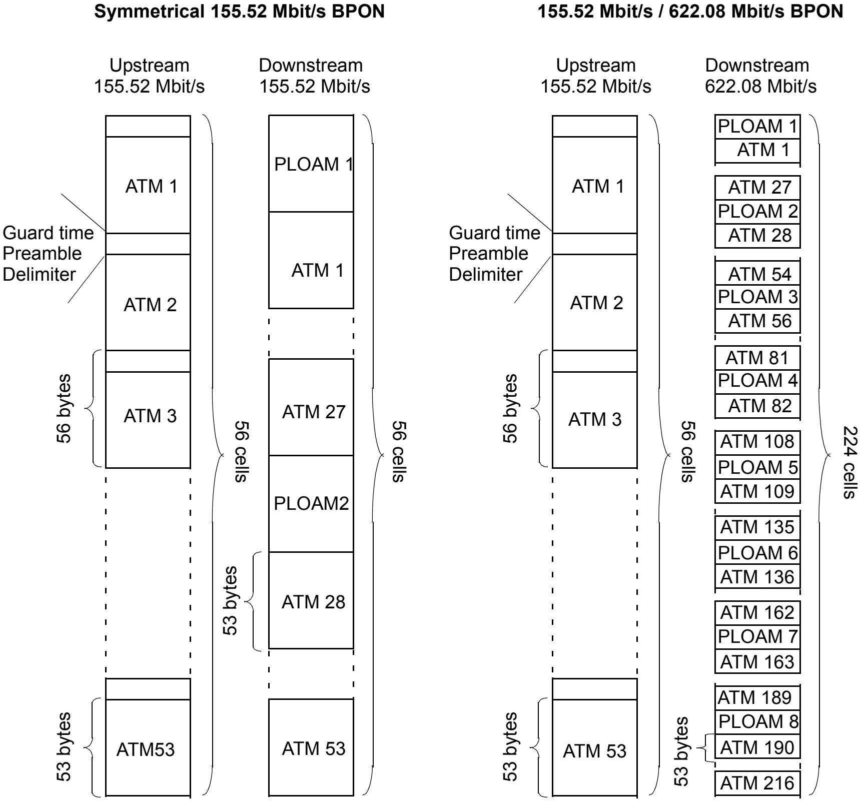
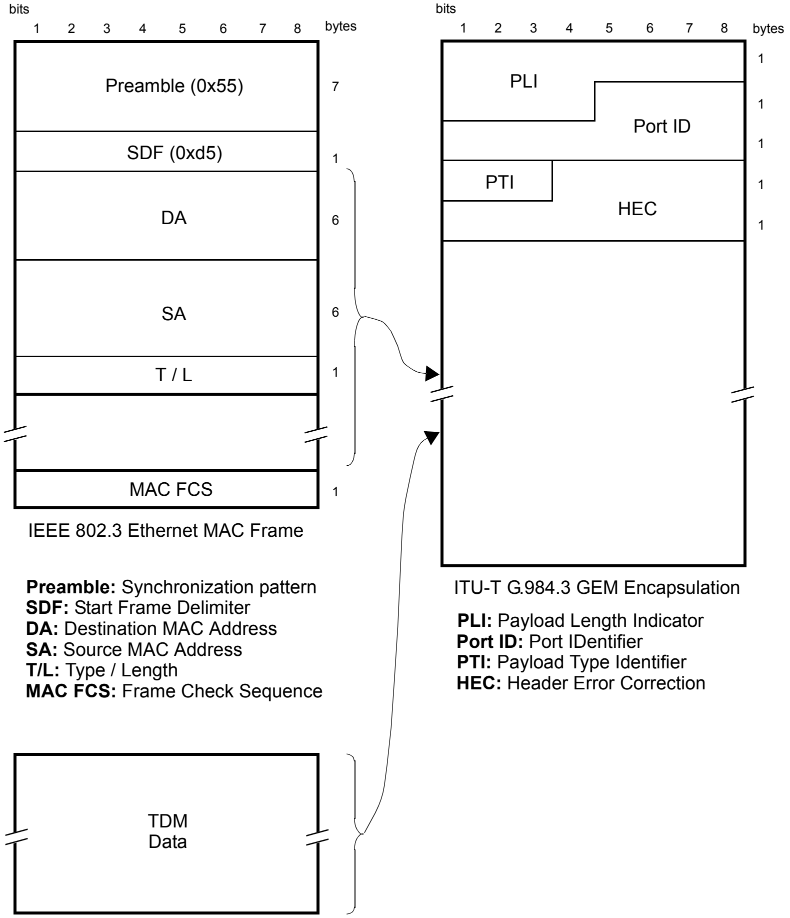


Figure 8.20 Upstream and downstream ATM cell flows in a BPON

head bytes to encapsulated data, and it can transport variable length packets. Packets can be encapsulated over a single GEM frame, or they can be fragmented and transported over various GEM frames. The GEM has two main objectives:

1. It provides delineation to encapsulated data. To do that, the GEM header contains a 12-bit *Payload Length Indicator* (PLI) that shows to the receiver where the next GEM frame starts. The 15-bit *Header Error Control* (HEC) field detects and corrects errors in the 5-byte GEM header, but it also helps with delineation.
2. It enables multiplexing of various data sources. These data streams are identified by means of a 12-bit label called Port ID.

Although GEM is designed to be the main encapsulation for GPON, ATM is also possible. GPON framing is designed to carry GEM encapsulated data and ATM simultaneously. The GPON downstream is made up of a continuous flow of 125 μs



Preamble: Synchronization pattern
SDF: Start Frame Delimiter
DA: Destination MAC Address
SA: Source MAC Address
T/L: Type / Length
MAC FCS: Frame Check Sequence

PLI: Payload Length Indicator
Port ID: Port IDentifier
PTI: Payload Type Identifier
HEC: Header Error Correction

Figure 8.21 The GEM encapsulation, as described in Recommendation G.984.3, enables transport of packetized data as well as TDM.

frames. A GPON downstream frame includes a *Payload Control Block* (PCBd), an ATM partition that contains ATM cells delivered to the ONUs, and a GEM partition with GEM encapsulated data (see Figure 8.22). The PCB provides different types of overhead for delineation, error detection, PLOAM and other purposes. The PCB also contains a downstream bandwidth map with a pointer list that indicates the time at which each ONU may begin and end its upstream transmission. The upstream is made up of multiplexed transmission bursts from the different ONUs. Every burst starts with a *Physical Layer Overhead* (PLO) that contains fields for delineation, error detection, transmitter identification by means an ONU-ID, and for other purposes. Several contiguous transmissions from the same ONU may be granted. If this happens, the data can be chained in the same burst under the same PLO header.

Besides the PLO, the upstream may contain other overheads by request of the ONUs:

- The PLOAM is a 13-byte overhead containing OAM messages. These messages usually carry alarms, information about the transmitter and encryption passwords and keys.
- The 120-byte *Power Leveling Sequence* (PLS) is inserted in the upstream to deal with the near–far problem typical of PON. The distance between the OLT and each ONU may be different, and therefore the power received by the OLT from the different ONUs varies. This may affect the ability of the OLT to decode the upstream without errors. The objective of the PLS is to help the OLT to solve this problem.
- The *Dynamic Bandwidth Report* (DBR), may have 1, 2 or 4 bytes of length. Messages sent in this overhead help the OLT to schedule the transmissions sent by the ONU.

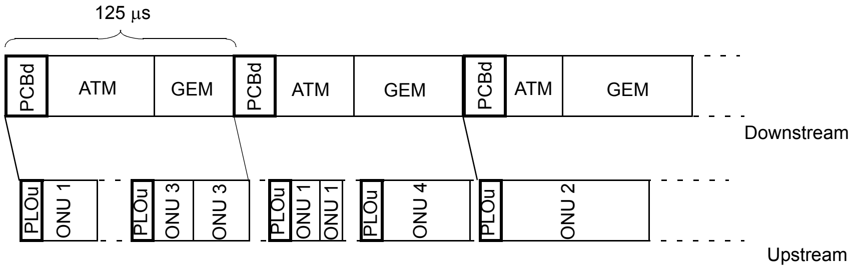


Figure 8.22 Upstream and downstream flows in a GPON. The downstream is composed of a continuous frame stream with ATM and GEM encapsulated data. The upstream contains data bursts from the different ONUs as they have been granted by the OLT in the downstream.

8.2.5 Ethernet PON

The Ethernet PON or EPON is the IEEE alternative for PON. The first version of EPON was released in 2004, which makes this technology the latest PON version to appear at the time of writing. EPON is based on Ethernet, the most successful networking technology specified by the IEEE. In fact, EPON is part of the EFM initiative that attempts to extend Ethernet to the local loop. EPON is a direct competitor of the GPON technology defined by the ITU-T.

There are two alternative interfaces for EPON, known as 1000BASE-PX10 and 1000BASE-PX20. The former has a minimum range of 10 km and the latter 20 km. The typical number of ONUs in an EPON is 16, but alternative splitting ratios are also possible. There is a trade-off between range and splitting ratio, because optical loss increases with both distance and split count. This means that more ONUs can be served if the distance between the ONU and the OLT is shorter.

All the currently defined EPON interfaces are for transmission at 1 Gbit/s, but a 10-Gbit/s EPON standard is expected to be available soon.

The EPON upstream and downstream are duplexed in a single SMF fiber. The upstream is transmitted at a nominal wavelength of 1310 nm, and the downstream at 1490 nm. This allows for the EPON to coexist with other services, such as broadcast video or private DWDM transmitted in the 1550 nm window. The signal is encoded with the same 8B/10B code that was specified by most of the Gigabit Ethernet interfaces operating at 1 Gbit/s. This means that the signaling rate for 1 Gbit/s EPON is 1.25 GBd.

The main goal of the 1000BASE-PX physical interfaces is to provide an access point where to connect MAC entities capable of transmitting standard IEEE 802.3 MAC frames. PON networks are a mixture of a dedicated and shared medium and EPON emulates point-to-point links over this medium. To do that, it extends the traditional Ethernet physical layer by defining:

- A scheduling protocol called *Multi-Point Control Protocol* (MPCP) that distributes transmission time among the ONUs to avoid upstream traffic collisions.
- Tags known as *Logical Link Identifiers* (LLID) that define point-to-point associations between the ONU and the OLT at physical level.

As a result, the EPON is compatible with most of the advantages provided by switched Ethernet networks like IEEE 802.1D bridging or VLANs. These features can be provided by the ONUs and OLTs themselves. Furthermore, the EPON defines other features that are not native in traditional Ethernet networks. For example, *Forward Error Correction* (FEC) is defined to increase range and splitting ratio.

8.2.5.1 The Multipoint Control Protocol

The *Multi-Point Control Protocol* or MPCP is a signaling protocol for EPON, and it's main function is to allow the OLT to manage the downstream bandwidth assigned to the ONUs. This protocol can perform other functions as well, namely:

- Enable the ONUs to request upstream bandwidth for transmission, and the OLT to assign this bandwidth in a way that collisions do not occur and network utilization is optimized.
- Allow parameter negotiation through the EPON network.
- Enable ranging by monitoring the *Round Trip Delay* (RTD) between ONUs and OLT. This feature is important for correctly scheduling upstream transmissions.
- Support ONU autodiscovery and registration.

The MPCP is implemented as an extension of the MAC control protocol and therefore MPCP messages are carried over standard Ethernet frames with the Type/Length field set to 0x88-08. There are five MPCP messages currently defined.

- GATE – grants access to the upstream bandwidth for the ONUs for certain periods of time.
- REPORT – used by the ONUs to report local information to the OLT. This information is used by the OLT to decide how the upstream bandwidth is distributed.
- REGISTER, REGISTER_REQUEST and REGUISTER_ACK – used for registering ONUs in the network.

The IEEE standards define the protocol for scheduling bandwidth, but equipment manufacturers select the actual scheduling algorithm.

8.2.5.2 Logical Link Identifiers

Logical Link Identifiers or LLIDs are physical layer link identifiers defined to enable 802.1D bridging over an EPON (see Figure 8.23). The LLID is delivered in EPON Ethernet frames as a 16-bit field that replaces the two last bytes of the frame preamble (see Figure 8.24). This field is added when a frame is transmitted by an EPON interface and transparently removed when received before being processed by the MAC layer.

LLIDs define point-to-point associations or logical links between the ONU and the OLT. Link identifiers are dynamically assigned when ONUs are registered in OLTs as a part of the initialization process. ONUs and OLTs choose the LLID to put

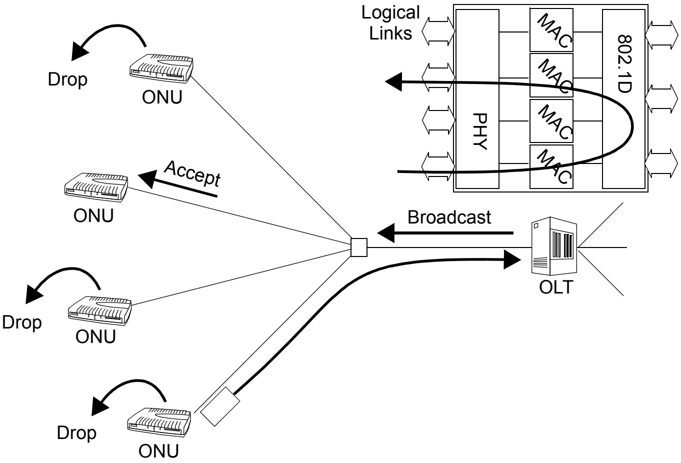
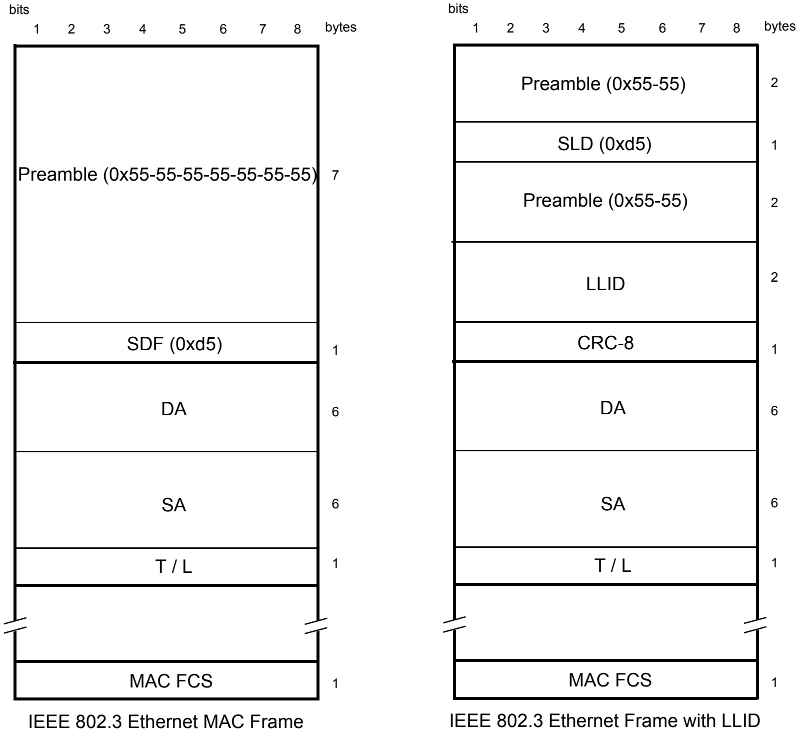


Figure 8.23 ONU-to-ONU bridging would not be possible without LLIDs. ONU-to-OLT associations defined by the LLIDs can be considered as point-to-point logical links. An 802.1D bridge can then perform learning and forwarding operations on the logical links.

in the delivered frames depending on the logical link they wish to use. Point-to-point emulation is achieved by following simple filtering rule: If a frame is received by an ONU or OLT with an LLID matching a known link identifier, it is forwarded to the right MAC entity that processes it. Otherwise, the frame is discarded. ONUs need to support a single LLID. They mark outgoing frames with the LLID assigned to them, and they accept frames marked with this LLID. The OLTs are more complex: they need one LLID per connected ONU.

The point-to-point link emulation is the primary operation mode for EPONs, but they may optionally support a shared LAN emulation mode (see Figure 8.25). To do this, it is necessary to modify the LLID filtering rules. The ONU must now accept all the incoming frames, not just the ones marked with their own LLID. The OLT must accept all the incoming frames and reflect them back when they match the defined forwarding rules.

It is also possible to take advantage of the broadcast nature of the downstream by defining a special channel called *Single Copy Broadcast* (SCB) channel. Frames sent by the SBC channel are accepted by all the ONUs.



Preamble: Synchronization pattern
SDF: Start Frame Delimiter
DA: Destination MAC Address
SA: Source MAC Address
T/L: Length / Type
MAC FCS: Frame Check Sequence

SLD: Start of LLID Delimiter
LLID: Logical Link IDentifier
CRC-8: Cyclic Redundancy Check parity

Figure 8.24 The preamble of an Ethernet frame carries the LLID, the SLD that helps processing the modified frame, and a CRC that detects errors in these new fields.

8.3 Ethernet in the First Mile

The standard IEEE 802.3ah for *Ethernet in the First Mile* (EFM) was released with the aim of extending Ethernet to the local loop for both residential and business customers.

EFM interfaces provide low and medium speeds when compared with the available LAN or WAN standards (see Figure 8.26). The new interfaces, however, are optimized to be profitable in the existing and newly installed provider access net-

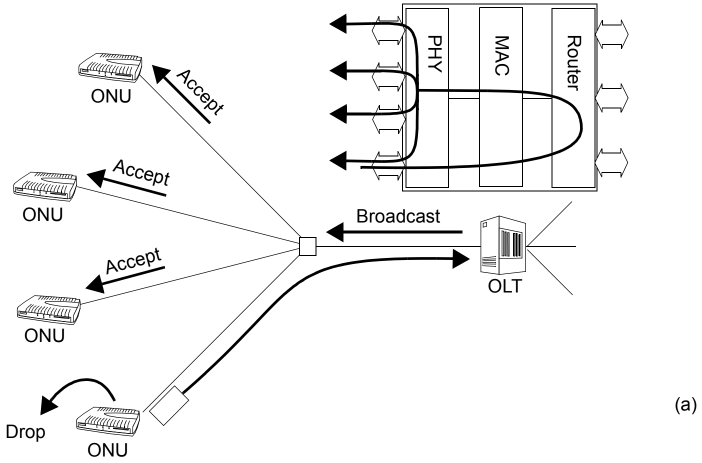


Figure 8.25 Optional shared LAN emulation performed by a PON. Frames are reflected by the OLT and accepted by all the ONUs except for the initial transmitter.

works. The copper EFM takes advantage of DSL technology for telephone copper pairs, and optical EFM is available for both PON networks and active Ethernet (see Table 8.4).

A major improvement provided by the IEEE 802.3ah is the definition of link *Operation, Administration and Maintenance* (OAM) services. The link OAM enables access network operators to monitor and troubleshoot the Ethernet link between the customer and network operator equipment. This new type of OAM complements OAM signalling at the service level defined in IEEE 802.1ag. The difference between IEEE 802.3ah and IEEE 802.1ag is that while the former works at the link level, the latter has been designed for OAM end-to-end signaling.

The functions of the link OAM protocol can be summarized as follows:

- *Discovery* – Identifies the devices at each end of the link, along with their OAM capabilities.
- *Link Monitoring* – Detects and indicates link faults, providing statistics on the registered errors.
- *Remote Failure Indication* – Reports a failure condition detected by the remote peer of a given switch, such as loss of signal in one direction of the link, an unrecoverable error, etc.
- *Remote Loopback* – Puts the remote peer of a given switch in loopback mode. When a switch is operating in loopback mode, it returns all the traffic it re-

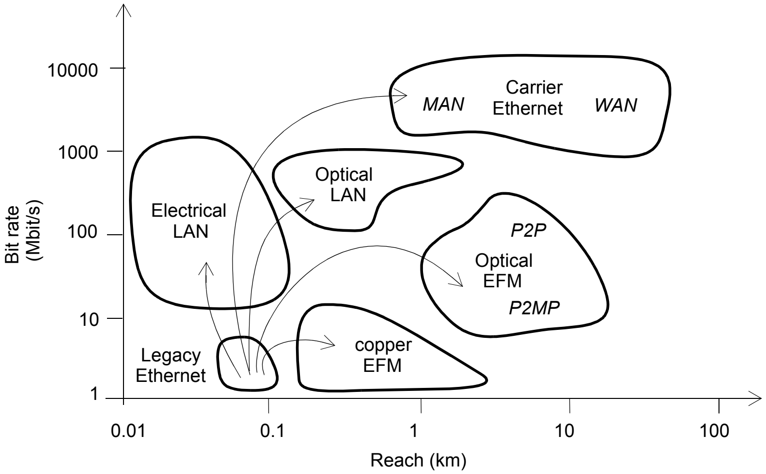


Figure 8.26 Ethernet applications and EFM

ceives back to the origin. The remote loopback mode is very useful for testing purposes.

Table 8.4
EFM Interface Summary

Interface	*Medium*	*Wavelength (nm)*	*Rate (Mbit/s)*	*Reach (km)*
100BASE-LX10	Two single-mode fibers	1310	100	10
100BASE-BX10	One single-mode fiber	1310 (US), 1550 (DS)	100	10
1000BASE-LX10	Two single-mode fiber	1310	1000	10
1000BASE-LX10	Two multimode fiber	1310	1000	0.55
1000BASE-BX10	One single-mode fiber	1310 (US), 1490 (DS)	1000	10
1000BASE-PX10	One single-mode fiber PON	1310 (US), 1490 (DS)	1000	10
1000BASE-PX20	One single-mode fiber PON	1310 (US), 1490 (DS)	1000	20
10PASS-TS	One or more telephone pairs	-	10	0.75
2BASE-TL	One or more telephone pairs	-	2	2.7

8.3.1 Ethernet over Telephone Copper Pairs

The EFM standard defines two interfaces for Ethernet transmission over telephone copper pairs:

- The 2BASE-TL interface is best suited to long-haul applications. It provides a symmetric, full-duplex 2-Mbit/s Ethernet transmission channel with a nominal reach of 2.7 km. It is based on SHDSL as per ITU-T G.991.2. The 2BASE-TL interface is optimized for local exchange applications.

- The 10PASS-TS interface is intended for short-haul applications. It offers a symmetric, full-duplex 10 Mbit/s transmission with a nominal reach of 750 m. It is based on the VDSL (ANSI T1.424) technology and optimized for deep fiber roll-outs like FTTB or FTTCab. It can be combined with EPON or active Ethernet to offer a simple bridged access network. The 10PASS-TS interface is compatible with baseband transmission of analog voice.

Although the EFM standard is expected to boost the use of Ethernet in provider access networks, Ethernet over copper pairs is really nothing new. The ITU-T ADSL2 standards defined in 2002 a new encapsulation for a *Packet Transfer Mode* (PTM) suitable for Ethernet frame transport. Later the PTM was added to other DSL standards like ADSL2+ and VDSL2. Before the release of ADSL2 and the PTM, Ethernet had to be transported by means the DSL ATM with the mappings defined by the RFC 2684.

The 10PASS-TS and 2BASE-TL are mostly based in existing technology, such as SHDSL and VDSL, mainly for the following reasons:

- Extensive DSL deployments exist and have existed for the past 10 years or so. DSL is a well-known technology, and network operators have a lot of experience with it.
- DSL has proven to be efficient, cost-effective and easy to deploy.
- National-level spectrum compatibility standards make it difficult to introduce signals with new spectrum shapes.

One of the challenges of Ethernet over copper is the lack of a strict definition of what is understood by a voice-grade copper pair. The reason for this is that telephone cabling started in the 19th century, much before any telecommunication regulations. Most of the current telephone pairs fall into the TIA / EIA categories 1 and 3 (see Table 8.5). Unlike other Ethernet standards, 2BASE-TL and 10PASS-TS are not specified for a transmission media of known features, and therefore the performance of these interfaces remains largely unpredictable in untested cables.

Table 8.5
Copper Pair Categories

Category	*Bandwidth*	*Common Application*
1	-	Telephony, ISDN BRI
2	4 MHz	4 Mbit/s Token Ring
3	16 MHz	Telephony, 10BASE-T, 100BASE-T4 (four wires)

Table 8.5
Copper Pair Categories

Category	*Bandwidth*	*Common Application*
4	20 MHz	16 Mbit/s Token Ring
5	100 MHz	100BASE-T, 1000BASE-T (four wires), short haul 155 Mbit/s ATM
5e	125 MHz	100BASE-T, 1000BASE-T (four wires), short haul 155 Mbit/s ATM
6	250 MHz	1000BASE-T (four wires)

One of the few changes introduced by the IEEE in the DSL specifications was the encapsulation defined for Ethernet. The original ITU-T encapsulation was based on an HDLC framing, but HDLC needs to add extra overhead bytes to avoid false alignment sequences within the frame. As a result, the length of an HDLC frame depends on its contents, and this makes it difficult to provide deterministic data bitrate. The HDLC encapsulation was replaced by the new 64/65-octet encapsulation. Finally, the 64/65-octet encapsulation has been accepted by the ITU-T, and it seems that it will be the main encapsulation for Ethernet over DSL in the near future.

The 64/65-octet encapsulation operates on 64-byte traffic fragments. It encodes both data and idle times. The traffic to be encoded is fragmented in 64-byte blocks. A single sync byte is added to the 64-byte fragments, producing a continuous stream of 65-byte blocks.

Another important feature of the EFM interface for copper is the *bonding function*. This feature is useful in providing Ethernet services over copper without the severe rate limitations. It can replace fiber in places where fiber is not available.

Ethernet bonding allows up to 32 pairs to be grouped together to make them appear as a single-capacity transmission channel (see Figure 8.27). Ethernet bonding works on heterogeneous links, since it is independent of the physical layer.

Bonding is possible even if the links operate at different speed. The IEEE 802.3ah bonding constitutes an Ethernet-optimized alternative to ATM-based bonding (see Paragraph 8.1.4).

8.3.2 Ethernet in Optical Access Networks

Optical EFM interfaces provide better performance than copper EFM in terms of reach and bit rate, but they require optical fiber. These interfaces have been especially developed for deep-fiber rollouts based on *Point-to-Point* (P2P) and *Point-to-MultiPoint* (P2MP) architectures.

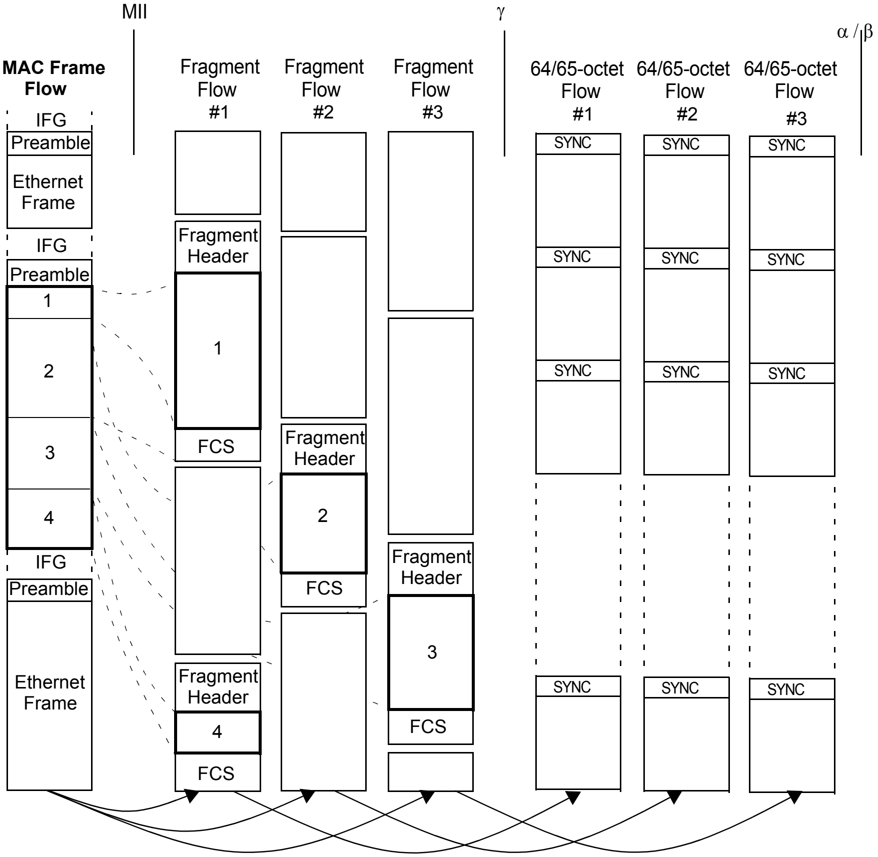


Figure 8.27 64/65-octet coding of an Ethernet-bonded interface made up of three physical links

In the case of the P2MP architecture, the EFM interface offers EPON (see Paragraph 8.2.5). For P2P, the EFM adapts the available Ethernet interfaces so that they operate in the access network. For example, bidirectional interfaces take advantage of the WDM technology to duplex the upstream and downstream in a single fiber. This makes it unnecessary to install two fibers per customer. Extended temperature operation is another improvement important for external plant operation.

The existing EPONs provide 1 Gbit/s symmetrical capacity, typically to be shared by 16 subscribers. This means that the minimum guaranteed bandwidth in FTTH is around 60 Mbit/s per subscriber, but depending on the network load, it

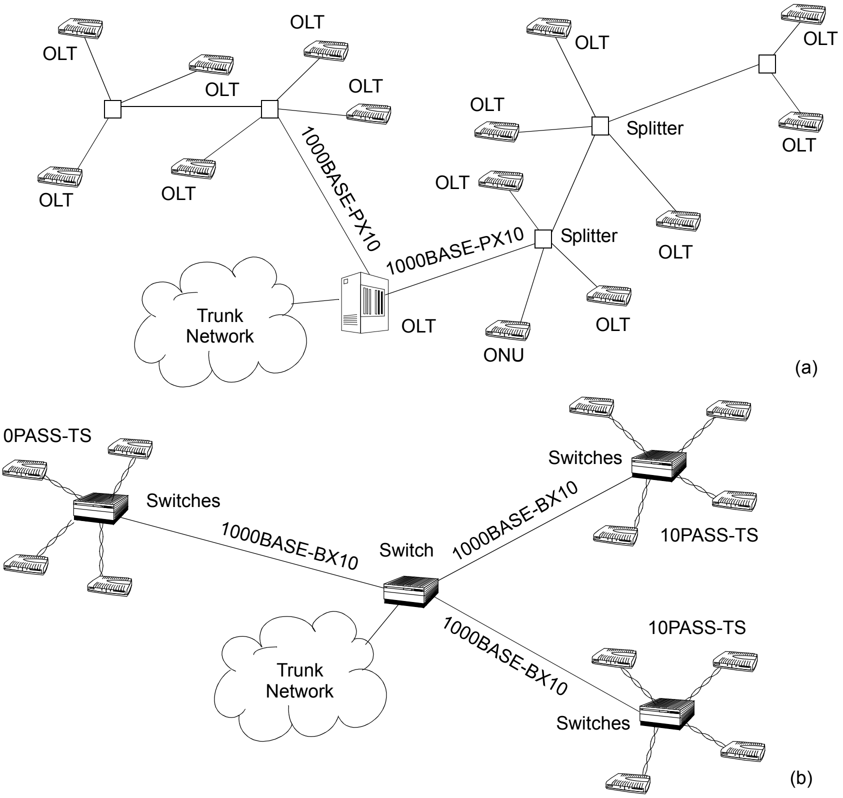


Figure 8.28 Two roll-out alternatives for residential customers based on EFM interfaces. (a) Deployment based on a shared EPON. (b) Deployment based on dedicated P2P optical links and short haul copper links

could increase up to several hundreds of megabits per second. The P2P interfaces for active Ethernet roll-outs provide 100 Mbit/s per customer in FTTH. Gigabit interfaces also exist, but these are typically used for backhaul in fiber-to-the-neighborhood applications, or they may be combined with copper in FTTB deployments (see Figure 8.28).

8.4 Service Provisioning

The legacy model for offering Internet access in DSL networks is quite simple. Connections are managed by the *Point-to-Point Protocol* (PPP). This protocol performs user authentication and IP address assignment. PPP connections are usually terminated in the *Remote Access Server* (RAS), a router managed by an ISP. In the local loop, IP data is encapsulated in an ATM *Permanent Virtual Circuit* (PVC). The metropolitan network is usually based on ATM or FR. Subscriber PVCs are forwarded to the ISP *Points of Presence* (PoP) by this network.

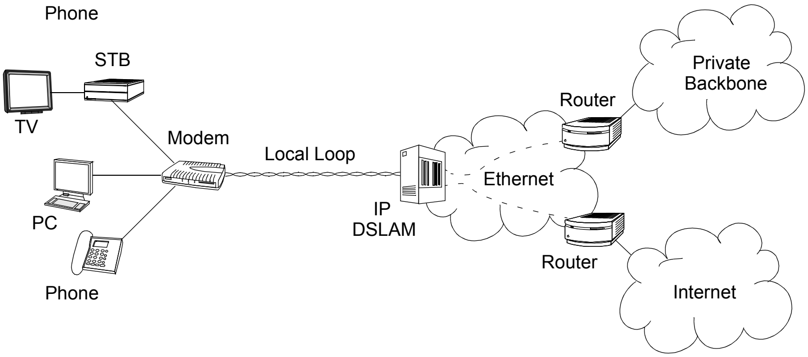


Figure 8.29 Network architecture for triple play service to residential subscribers. ATM and FR are replaced by Ethernet and IP.

The evolution of the access network makes it appear similar to a big LAN with multimedia capabilities (see Figure 8.29). This affects the service provisioning model in several ways, namely:

- The *Dynamic Host Configuration Protocol* (DHCP) tends to be the chosen protocol for session management instead of PPP. DHCP is better suited for non-connection-oriented technologies, such as Ethernet and IP. In order to provide security, DHCP must implement the relay agent identification option (option 82) defined in RFC 3046. With this option, the DHCP relay (usually the DSLAM) can forward information about itself and about its connection to the subscriber. This information makes it possible for the DHCP servers to implement security policies based on the port where the subscriber is connected instead of its MAC address (see Figure 8.30).
- ATM-based DSLAMs are being replaced by IP DSLAMs with Gigabit Ethernet interfaces, IEEE 802.1D bridging and IP routing features. IP DSLAMs are well suited for provision of IP multimedia services, and they are more cost-effective than ATM DSLAMs. Many of these new DSLAMs are being deployed out of the local exchange and closer to the subscribers as a part of deep-fiber roll-outs.
- ATM and FR switching are being replaced by Metro Ethernet bridging and *Multi-Protocol Label Switching* (MPLS). The new technology can provide point-to-point and multipoint-to-multipoint services of custom QoS.

- ATM encapsulation in the local loop is being replaced by Ethernet encapsulations like the one defined in the EFM standard.

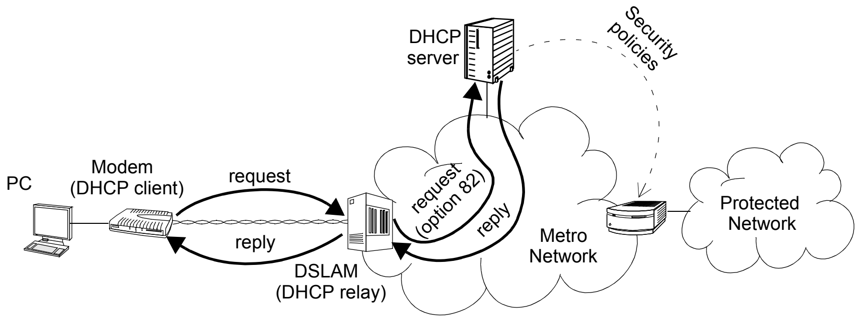


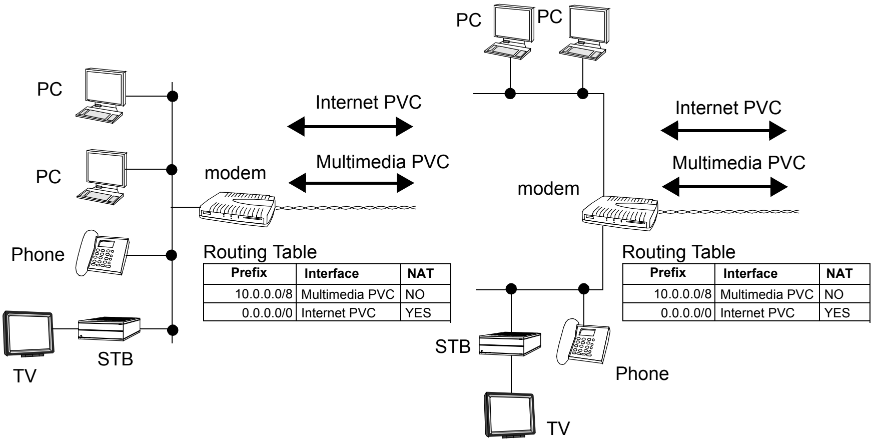
Figure 8.30 Usage example of DHCP with option 82 for providing connection parameters to users in a WAN. The DSLAM adds the option 82 to DHCP requests to inform the DHCP server on the port where the user is connected. This information is used by the DHCP server for providing the right information to that user and grant access to protected resources.

Since the current development of the Internet is not QoS-aware, many operators are developing their own private backbones to deliver IPTV, VoD and other QoS-sensible services. These private backbones are carefully designed IP networks with support for QoS by means of IS, DS, MPLS or other mechanisms. Using private addressing for this private network segregated from the Internet helps saving scarce public IPv4 addresses. The network becomes unroutable from the Internet, but this is not a problem, because it does not need to be part of the Internet. Access to private backbones for multimedia service provisioning can be solved by using several PVCs or VLANs to deliver different services and allocating one of them for best-effort Internet access. It is even possible to use different mechanisms for IP address assignation in different PVCs. For example PPP can be used in the Internet PVC, and DHCP in PVCs for multimedia applications.

Another advantage of segregating Internet and multimedia PVCs is that it is possible to disable *Network Address Translation* (NAT) only on the multimedia PVCs, thus improving transparency in service-provider applications while maintaining optimum usage of public addresses for the Internet access application. Disabling NAT is almost a need in the multimedia PVCs, because some protocols that can be potentially used in this interface may have problems passing through the NAT filter. This happens for example with the *Session Initiation Protocol* (SIP).

There are, however, certain issues if NAT is disabled in multimedia PVCs. The devices attached to the subscriber network become directly routable from the service provider private backbone. This means that the IP addresses of these devices must be unique, and they must be managed by the service provider rather than by the subscriber. When this approach is used, the service provider limits the maximum number of devices that can be connected to the subscriber LAN.

Some service providers deploy two different subnetworks for each subscriber. The first of them contains private addresses routable from the private backbone. This network is used for connecting multimedia devices. The second network has private addresses not routable from the private backbone, and it is used for Internet access (see Figure 8.31). With this second approach, the subscriber keeps full control over the Internet access subnetwork. The problem is that this solution makes it difficult to receive Internet-related contents in the multimedia subnetwork, or multimedia-related contents in the Internet access subnetwork.



Prefix	Interface	NAT
10.0.0.0/8	Multimedia PVC	NO
0.0.0.0/0	Internet PVC	YES

Prefix	Interface	NAT
10.0.0.0/8	Multimedia PVC	NO
0.0.0.0/0	Internet PVC	YES

Figure 8.31 Two options for the subscriber LAN: (a) Model based on a single network. Multimedia devices and PCs are connected to the network. (b) Model based on two subnetworks. One of them is for multimedia devices, the other for PCs.

Chapter 9

The Quadruple Play

The expression “quadruple play” refers to the commercial bundle of voice, video and data plus mobility. Quadruple play makes sense because mobile operators are aiming to add to their offer advanced data and video services. The voice-centric business model of mobile operators is becoming much more rich and complex. The objective is to provide similar services than wireline operators but over mobile terminals. Multimedia conferencing or mobile TV are seen by mobile operators as strategic services. Online banking, shopping or gaming are popular for wireline subscribers but they are becoming popular for wireless users as well.

On the other hand, fixed operators are looking for mechanisms to add mobility to their services. The indoor home connection to the Internet based on Wireless LAN (WLAN) rather than Ethernet or other wireline technology is now very common. Another example are mobile phones that become WLAN phones when a WLAN network is available.

Clearly, there have been two differentiated quadruple play strategies:

- *Migration of all services to mobile interfaces or Fixed Mobile Substitution (FMS).* The basis of this strategy is that users prefer mobile services if price and performance are of the same level than wireline. It is assumed that mobile technology evolution will enable high service level and specifically high bandwidth interfaces. Mobile access networks are often considered easier to deploy than wireline networks. However, price for operation over licensed radio bands and user equipment price could influence the price of mobile services. The all-over-mobile strategy is easier to follow by pure mobile operators without fixed facilities.
- *Integration of fixed and mobile business divisions or Fixed Mobile Convergence (FMC).* Integrated operators are able to provide services both over wireline and wireless interfaces. Ideally, the services would be the same without regard of the particular access network. Of course, different access networks may provide different performance level or may be better suited to certain services. The control plane will provide the mechanisms needed to adapt services

to the particularities of every access network. This strategy is good for companies that already have a mobile and a fixed business divisions. Many incumbent operators are examples of this.

Both the FMS and the FMC strategies require important shifts in business models and technology involving service definition, charging, roaming between operators, connection with service platforms and many others. The telecommunications industry is looking for the mechanisms that will allow interconnection of services and subscribers through an heterogeneous network made up by several wireline and wireless access technologies. Some of the most important initiatives are the standardization of an IP Multimedia Subsystem (IMS) by the 3rd Generation Partnership Project (3GPP) and the work about the Next Generation Network (NGN) led by the ITU-T and ETSI.

9.1 Cellular Communications Overview

Modern mobile communication networks are cellular. In such networks, mobile users are served by fixed radio transceivers called base stations. Base stations cover areas geographically limited but if all base stations are considered together, they are able to serve a geographically wide area. As users move, communications are transferred between different base stations in a process called handoff. Therefore, the handoff mechanisms constitute the basis of mobility in cellular networks.

The first commercial cellular communications systems date to the early 1980s. These 1G systems were based on an analog user plane and used different frequency bands to multiplex voice calls and allow multiple access. Many 1G mobile telephone systems were created in different countries, usually with very limited interoperability. Some examples of 1G mobile telephone systems are the Advanced Mobile Phone System (AMPS) for the American market, Total Access Communication System (TACS) for UK or the Nordic Mobile Telephone (NMT) for Northern Europe.

Analog 1G mobile telephone systems were replaced by 2G digital systems by the end of 1980s. The new generation brought more quality, capacity and security at lower cost. In the US, Canada, and other countries, AMPS evolved to the system known as Time Division Multiple Access (TDMA[1]) or Digital AMPS (DAMPS). The original standard for TDMA/DAMPS is the Interim Standard-54 (IS-54). The IS-54 was later improved by the IS-136. Mobile communications based on Code Division Multiple Access (CDMA) were introduced by Qualcomm more or less at the same time that TDMA/DAMPS and both technologies became competitors. The CDMA standard IS-95 was approved in July 2003 by the Telecommunications In-

[1] The TDMA/DAMPS takes its name from the TDMA multiple access technology but both concepts must not be confused.

dustry Association (TIA) and has been marketed with the branch name of cdmaOne in the US and a number of other countries in North America, Latin America, and Asia. CDMA enables multiple access by associating to different users, different and orthogonal code sequences rather than using time sequences like in TDMA or different slices of the frequency spectrum like it's done with Frequency Division Multiple Access (FDMA). Traffic from every user in a CDMA system can be recovered thanks to the particular properties of orthogonal sequences. Since its introduction for commercial mobile communications in the 1990's, CDMA represented a success due to the better performance exhibited when compared with its other 2G competitors. Particularly, cdmaOne systems brought broader coverage, better spectrum usage and lower power emission. Thanks to it's features, CDMA technology is used in 3G mobile networks as well.

The Japanese 2G mobile telephony standard is the Personal Digital Cellular (PDC) system. The PDC standard was released in 1991 and NTT DoCoMo launched the first network based in PDC technology in 1993. PDC operates at the 800 and 1500 MHz bands and uses TDMA multiple access technology. PDC was successful only in Japan. After peaking at 80 million lines it is currently becoming obsoleted by 3G technologies.

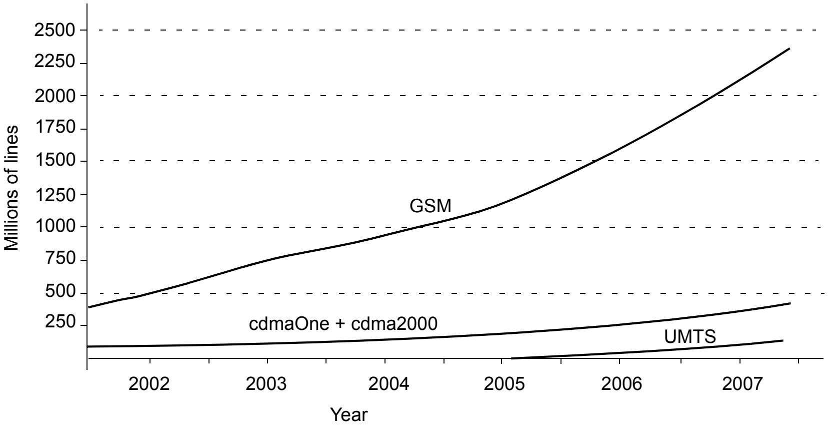


Figure 9.1 Evolution of the worldwide cellular telephony market

However, the most successful 2G mobile communications system was developed in Europe. In 1982, the *Conférence Européenne des aministrations des Postes et des Télécommunicacions*, create the *Groupe Spécial Mobile* (GSM) with the objective of developing an European standard for mobile telephony. Later, in 1989, this responsibility was transferred to the recently created European Telecommunications Standards Institute (ETSI). In 1990 the ETSI published the phase I of the GSM specifications and the first commercial GSM service saw the light in 1991 in Finland. Since then, the number of GSM subscribers and networks has been growing steadily. Today, GSM has become almost the *de facto* standard for mobile communications and there are GSM networks in most of the countries. The number of GSM connections is approaching to 2,500 billion (see Figure 9.1) and four of every five mobile phones have GSM connectivity. These are the reasons because GSM has been renamed to *Global System for Mobile Communications*.

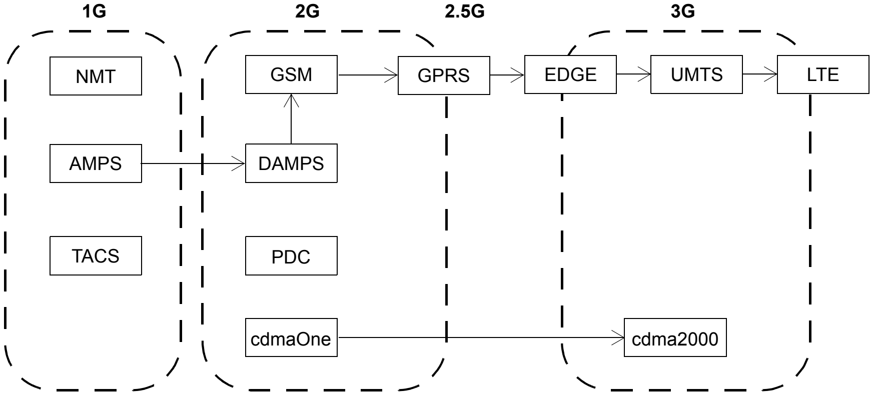


Figure 9.2 Evolution of mobile communications standards

The driver for bringing into the market 3G mobile systems was demand of multimedia and data services for mobile devices. A global standard for 3G mobile systems was unsuccessfully sought in the late 1990's. Finally, the ITU-T standard for 3G mobile communications, known as International Mobile Communications 2000 (IMT-2000) accepted five different radio interfaces:

- *IMT-2000 CDMA Direct Spread*, also known as Wideband CDMA (WCDMA), this radio interface constitutes the basis of the 3G Universal Mobile Terrestrial System (UMTS)
- *IMT-2000 CDMA Multi-Carrier*, the radio interface for the cdma2000 system, successor of Qualcomm cdmaOne.
- *IMT-2000 CDMA Time Division Duplexing*, is an alternative interface to WCDMA for the UMTS system. There are two variations known as Time Division CDMA (TD-CDMA) and the Time Division Synchronous CDMA (TD-SCDMA). The latter has been chosen as the air interface for 3G mobile systems in the People's Republic of China.
- *IMT-2000 TDMA Single-Carrier*, constitutes the basis of the Enhanced Data rate for Global Evolution (EDGE) system.

- *IMT-2000 FDMA/TDMA*, the air interface of the Digital Enhanced Cordless Telecommunications (DECT) system. This system is suitable for wireless voice and data communications with range of up to several hundreds of meters.

Inclusion of a sixth radio interface based on Orthogonal Frequency Division Multiplexing (OFDM) technology is under discussion. This fact would open the door to OFDM-based technologies like WiMAX to get spectrum allocations from the ITU-R and would allow these technologies to get a strong position in the definition of future mobile systems.

In practice, two important standards for 3G mobile communications systems arose from the IMT-2000 initiative: The 3GPP UMTS and the cdma2000 defined by the 3GPP2. UMTS takes GSM as a starting point. Some of the important features of UMTS are a new radio access network for increased bit rates in the air interface and a multiservice packet switched core. On the other hand, cdma2000 is the 3G successor of cdmaOne and it keeps the IS-95 1.25 MHz radio channel structure in all its variations:

- *cdma2000 1x*, is in fact a not a true 3G technology. It is rather referred as a 2.5G mobile system. Performance was limited to symmetrical peak rates of 144 kb/s in the first release but it was later improved to 307 kb/s
- *cdma2000 1xEV-DO*, where EV-DO means “EVolution - Data Only” or “EVolution - Data Optimized”. EV-DO is a true 3G technology. Initially it provided peak rates of 2.4 Mb/s in the downlink and 153 kb/s in the uplink (later it improved to a maximum in the 3.1 Mb/s in the downlink and 1.8 Mb/s in the uplink). The newest EV-DO release (Revision B) improves performance by means aggregation of multiple radio channels in a single traffic channel.
- *cdma2000 1xEV-DV*, where EV-DV means “EVolution - Data and Voice” was designed to replace EV-DO but it has not got enough support from vendors and carriers so far.

9.1.1 The Global System for Mobile Communications

The advantages of the GSM telephony system over 1G technologies and other 2G technologies are a good quality voice service, attractive prices, the Short Message Service (SMS) text-based application and extensive roaming services with many operators around the world.

9.1.1.1 Overview of the air interface

The GSM air interface, know as U_m, is a radio link between mobile stations and base stations. In most GSM networks, these radio links operate either in the 900 MHz or in the 1800 MHz bands. But in some American countries, including the US, the

850 MHz and the 1900 MHz are used instead. There are also some GSM networks operating in the 450 MHz band working in Scandinavia, Eastern Europe and Russia (see Figure 9.3).

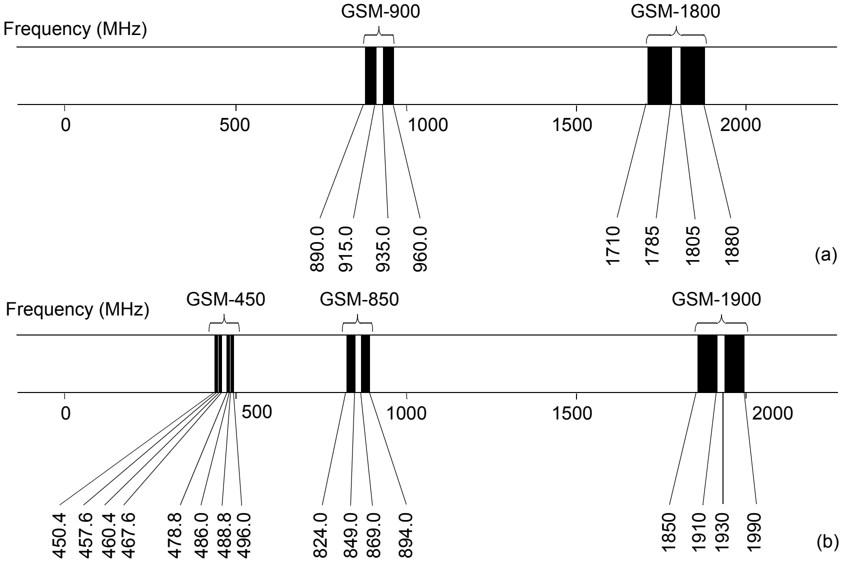


Figure 9.3 Frequency allocations of GSM cellular telephony system. (a) Common GSM-900 and GSM-1800 systems. (b) Less common GSM-450, GSM-850 and GSM-1900.

The basic GSM system uses two 25 MHz bands placed between 890.0 and 915.0 MHz for the uplink (link between the mobile station and the base station) and 935.0 and 960.0 MHz for the downlink (link between the base station and the mobile station). Each 25 MHz band contains 124 radio channels with 200 kHz carrier spacing. A full duplex radio channel is made up by an uplink and a downlink carrier pair. The GSM systems operating at 1800 MHz have 75 MHz for the uplink and 75 MHz for the downlink with 374 pairs of radio channels. However, not all carriers can be used simultaneously by every base station in the system. To avoid interferences only subsets of carriers can be assigned every base station (see Figure 9.4).

Every radio channel in the GSM system transports an 8-slot Time Division Multiplexing (TDM) frame. Each time interval in this frame is accessed by users with TDMA techniques and each radio channel in the GSM spectrum is accessed using FDMA. Multiple access in GSM is therefore a combination of FDMA and TDMA.

The 8-slot TDM frame of the GSM system transport Traffic CHannels (TCH), Slow Associated Control CHannels (SACCH), and sometimes Fast Associated Control CHannels (FACCH) in a multiframe structure. The user bits contained in one TDMA time slot are called burst. A burst contains 148 bits but only two groups of

57 bits are usable. TDMA bursts are separated by guard times equivalent to 8.25 bits to prevent collisions (Figure 9.5). A TCH can place up to 22.8 kb/s of user data but there also exists a half rate TCH operating at 11.4 kb/s. Even working at 22.8 kb/s, a TCH can not transport directly an ISDN 64 kb/s speech signal. This problem is solved with the help of voice compression algorithms that reduce speech bit rate to a maximum of 13 kb/s and provide reasonable quality. Before being transmitted, the encoded voice is protected against bit errors with redundancy bits and the bitrate is increased from 13 kb/s to 22.8 kb/s.

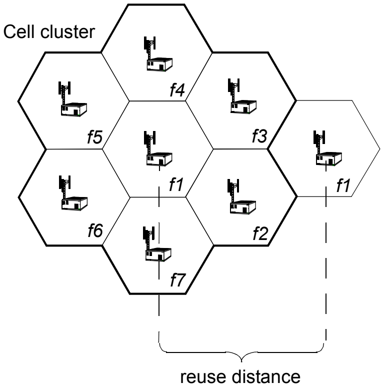


Figure 9.4 GSM cellular radio network structure. To minimize interferences not all the available carriers are used simultaneously. In the example, the 124 available pairs of carriers are distributed in seven sets of 17 frequency each. Carrier sets are assigned to the base stations of a seven-cell cluster. The structure of the cluster is then extender over a plane region of arbitrary dimensions. The maximum simultaneous number of calls this radio system can carry is 136 (17 x 8).

The GSM modulation is the Gaussian Minimum Shift Keying (GMSK). The GMSK is a single carrier, phase digital modulation. In GSMK, the modulating signal is pre-filtered with a filter with gaussian impulse response. The GSMK provides good spectrum efficiency (about 1 b/Hz), a compact spectrum with low radiation in adjacent radio bands, and a constant envelop signal in the time domain that allows the use of simple and efficient power amplifiers.

9.1.1.2 The GSM network

A GSM network is made up by the mobile stations carried by users, the base station subsystem that controls the radio links between the network and mobile stations, and the network subsystem that handles call switching and routing like any other telephone network and manages mobility of user equipment.

The base station subsystem is made up by two types of network elements:

1. The *Base Transceiver Station (BTS)* contains radio transmitters and receivers and implements the U_m interface to exchange information with mobile stations using radio waves.
2. A *Base Station Controller (BSC)* controls the operation of one or several BTSs. BSCs are connected to BTSs through the A_{bis} interface and to the network subsystem trough the *A* interface. The BSC allocates bearer resources for mobile stations and multiplexes low rate voice calls over high capacity link with the network subsystem. The BSC also coordinates handoffs of mobile stations if the involved BTSs are all directly connected to it and collects information about radio channels that helps determining when handoffs must start. Finally, the BSC performs interworking functions between the signaling protocols of the base station subsystem and the network subsystem.

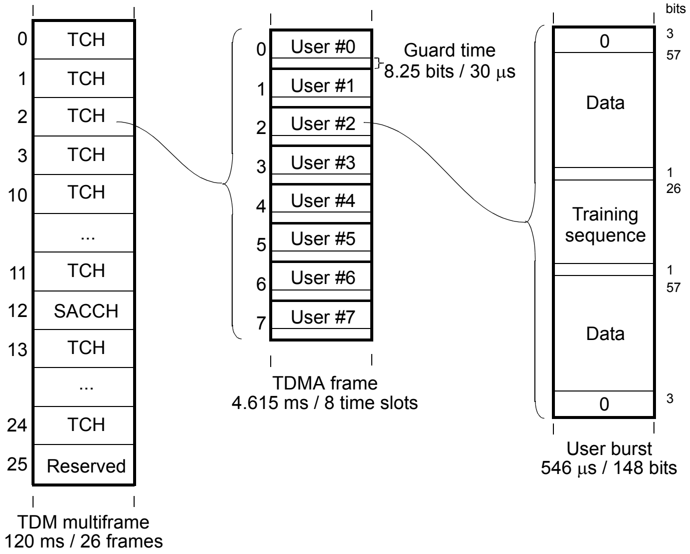


Figure 9.5 Simplified GSM framing structure

The core element of the GSM network subsystem is the Mobile Switching Centre (MSC). The MSC is linked to groups of BSCs and may also be connected to other MSCs thus providing global connectivity to the mobile network. MSCs are also gateways between the mobile network and the PSTN or the ISDN to enable calls between fixed and mobile users. An MSC behaves like a traditional telephone exchange in the sense that it can route calls to the destination using the telephone number of the destination. However, MSCs are more than PSTN/ISDN exchanges because they are in charge of managing mobility of user terminals. To do that, MSCs receive the support of different databases:

- The *Home Location Register (HLR)* maintains and updates mobile subscribers' profiles and current locations. Subscribers' location in GSM networks are maintained in real-time by means paging procedures specifically defined for this purpose.
- The *Visiting Location Register (VLR)* maintains profiles and locations of roaming subscribers in the visited network. Usually, the VLR is an integrating part of the MSC but this not need to be the case. A mobile user is said to be roaming if it is being served by a network other than their home network. Roaming GSM subscribers can use other operator's network if there exist an agreement with the home operator.
- The *Equipment Identity Register (EIR)* contains subscribers' equipment identities. It is useful to grant, limit or deny access to unauthorized mobile terminals.
- The *Authentication Center (AC)* provides keys and algorithms to maintain security in the network.

When the MSC needs to route a call it queries an HLR with the destination telephone number. The HLR then provides to the MSC the appropriate re-routing information. For roaming users, the HLR responds with the Mobile Station Roaming Number (MSRN) allocated for the user by the visited network. The MSRN contains the routing information necessary to extend the call from the home to the visited MSC. The visited MSC then contacts to the destination mobile station.

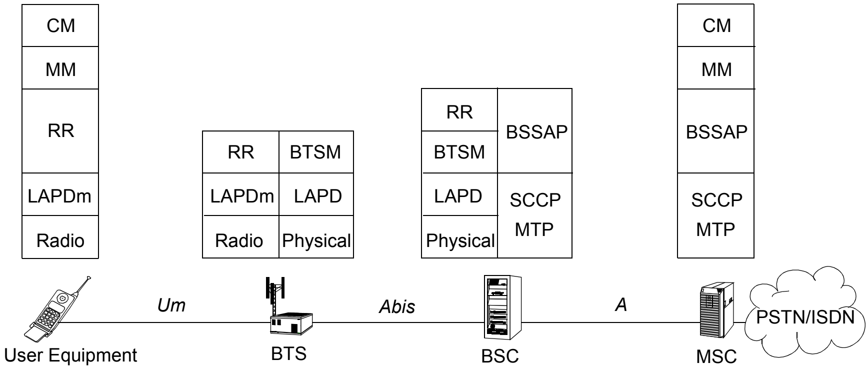


Figure 9.6 The GSM signaling protocol architecture is based on ISDN and SS7 protocols modified or extended when necessary

GSM was defined to be a mobile extension of the ISDN network and therefore many service definitions, interfaces, and protocols are taken from ISDN. signaling in GSM networks is strongly based on the ISDN and the Signaling System 7 (SS7) (see Figure 9.6). The GSM signaling architecture is structured in three layers. Layer 1 and 2 are equivalent to the Open Systems Interconnection (OSI) physical and data link layers. Layer 3 transports packetized signaling messages. The physical layer in the link between the user equipment and the BTS is the GSM FDMA/TDMA air interface. Other physical interfaces in the network are usually implemented with E1 trunks or other TDM interfaces. The data link layer in the radio interface is based on an adapted version of the ISDN Link Access Procedure - channel D (LAPD) known as LAPD modified (LAPDm). In the A_{bis} reference point (link between the BTS and the BSC), LAPDm frames are translated to normal LAPD frames. Physical connectivity, error detection and retransmission, message segmentation and reassembly, addressing, and message multiplexing in the *A* reference point (link between the BSC and the MSC) are provided by the Message Transfer Part (MTP) and the Signaling Connection Control Part (SCCP) SS7 protocols. The message layer is made up by three sublayers:

1. The *Radio Resource (RR)* management sublayer establishes and maintains a communications path between the user equipment and the MSC over which signaling and user data is delivered. Handoffs between cells are coordinated using the RR sublayer. The RR sublayer is terminated at the BSC but some functions may be delegated to the BTS. Once interpreted and processed by the BSC, RR information can be delivered to the MSC using the Base Station Subsystem Management Application Part (BSSMAP) protocol
2. The *Mobility Management (MM)* sublayer is relayed transparently by the BSC and terminated at the MSC. This sublayer provides miscellaneous services related with mobility management including mobile station location procedures along with authentication and ciphering.
3. The *Connection Management (CM)* sublayer is also relayed by the BSC to the MSC. It supports call control that manages circuit switched services such as voice calls, supplementary services management, and Short Message Service (SMS) delivery.

9.1.1.3 Transition to UMTS

Smooth migration to a 3G data-centric mobile network from the voice-centric GSM requires different evolution stages with progressive introduction of new applications and higher speed access.

GSM is optimized for circuit-switched voice service delivery and original data services for GSM were also circuit-switched. Specifically, the first data service for GSM was Circuit-Switched Data (CSD). This service provides a symmetrical 9.6 kb/s bitrate over a single GSM TCH. The 9.6 kb/s is even smaller than the 13 kb/s provided for voice due to the stronger error protection required for data services. The High Speed CSD (HSCSD), approved by ETSI in 1997, was an improvement to the CSD that allowed bit rates of several tens of kb/s by means two new features:

1. A *flexible time slot allocation* scheme that allows multislot configurations. Some popular configurations are the symmetric service made up with two slots for the downlink and two for the uplink and the asymmetric service with three slots for the downlink and one for the uplink.
2. A new channel encoding that allows 14.4 kb/s per slot rather than 9.6 kb/s per slot. However the 14.4 kb/s encoding is not as robust as the 9.6 kb/s and some mobile units may need to switch to the 9.6 kb/s mode under heavy interferences or noise.

Both CSD and HSCSD are circuit-switched and therefore they keep busy network resources while the communication lasts even if transmission is discontinuous. This situation led the telecommunication industry to look for packet switching technologies, much better suited for data services, for cellular networks. The packet-switched evolution of CSD/HSCSD is the General Packet Radio Service (GPRS). GPRS was specifically designed by ETSI for GSM networks but it as been integrated to DAMPS as well. GPRS greatly improves and simplifies wireless access to packet data networks like the Internet. Packet switching capabilities makes GPRS resemble a 3G technology but speed limitations in the radio access network makes GPRS pre-3G. For this reasons, GPRS is sometimes referred as a 2.5G technology. The advantages of GPRS can be summarized in three points:

1. *Better usage* of radio resources in data applications thanks to statistical multiplexing.
2. *Shorter access times* to the data service form the mobile stations. Usually in the range of one second or less. With CSD/HSCSD the access time used to be of several seconds.
3. *Flexible billing* based on the amount of transmitted data rather than in connection duration like in CSD/HSCSD.

Like HSCSD, GPRS uses flexible time slot allocation but GPRS allows statistic multiplexing of data from various users over the allocated space. One to eight slots can be allocated for GPRS. The uplink and the downlink are allocated separately and asymmetric configurations are allowed. Within the same radio channel, transmission capacity is dynamically shared between GPRS and circuit-switched services. Current traffic load and service priorities may be used to reallocate capacity periodically. Multiple access in the uplink is accomplished by individual grant messages delivered from the network to the mobile stations.

GPRS has its own channel coding mechanisms. In fact there are four different channel encoding schemes that provide different protection levels. CS-1 provides 9.02 kb/s per time slot, CS-2 provides 13.4 kb/s, CS-3 provides 15.6 kb/s, and CS-4 provides 21.4 kb/s. CS-4 is the fastest but it cannot be only used under good transmission conditions. Under heavy interferences or noise, it could be necessary to switch to CS-3, CS-2 or CS-1. The maximum theoretical bitrate provided by GPRS in one radio channel is 172.2 kb/s (8 times 21.4 kb/s) to be shared among all mobile stations using the same radio channel. In practical situations, however, performance is limited to a few tens of kb/s per user.

GPRS can be easily integrated in the GSM network. Requirements are fulfilled mainly by software upgrades of existing GSM network nodes and deployment of two new types of network elements usually referred as GPRS Support Nodes (GSN) (see Figure 9.7):

1. The *Serving GPRS Support Node (SGSN)* is responsible for the delivery of data packets from and to the mobile stations within its service area. Its tasks include packet routing and transfer, mobility management, logical link management, and authentication, authorization and charging of mobile users.
2. The *Gateway GPRS Support Node (GGSN)* is an interface between the GPRS backbone and external (IP, X.25,...) packet data networks. It translates between GPRS packet formats and the Packet Data Protocol (PDP) format corresponding to the external network. In the GGSN, PDP addresses are converted to GSM identifiers.

The SGSN and the GGSN are connected via a GPRS backbone that is IP based. Packets traversing the GPRS backbone are encapsulated and tunneled using the GPRS Tunneling Protocol (GTP).

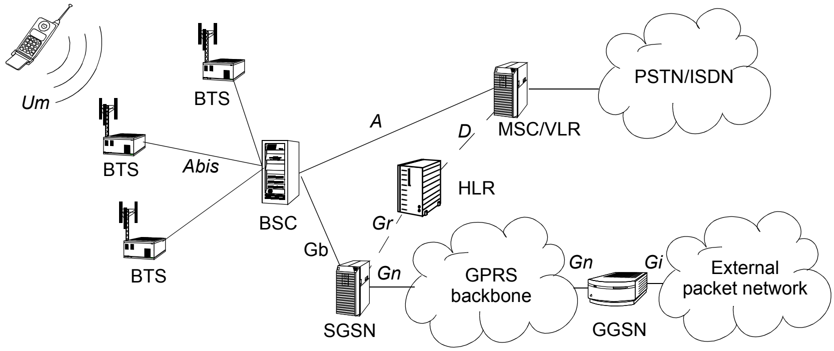


Figure 9.7 Simplified view of the 2.5G GPRS mobile network.

Mobile stations may have three different levels of support of GPRS. Class A devices support simultaneous operation of GPRS and conventional GSM services. Class B devices support GPRS and conventional GSM services but it can only use one of them at a given time. Finally, class C devices cannot register simultaneously to GPRS and GSM networks.

GPRS users are registered in a SGSN before accessing any service provided by an external network. The SGSN, authenticates users and assigns them a Packet Temporary Mobile Subscriber Identity (P-TMSI). To exchange information with external packet networks, a GPRS mobile station must also be attached to one or more addresses from these networks. For example, if the mobile user wants to communicate with an IP network it will need an IP address. GPRS users get the data they need to operate with external networks when they start a session. This procedure is known as PDP context activation procedure. During the PDP context activation procedure, the network assigns a PDP address to the user but also other parameters like a QoS profile and the address of a GGSN to route packets to the external network.'

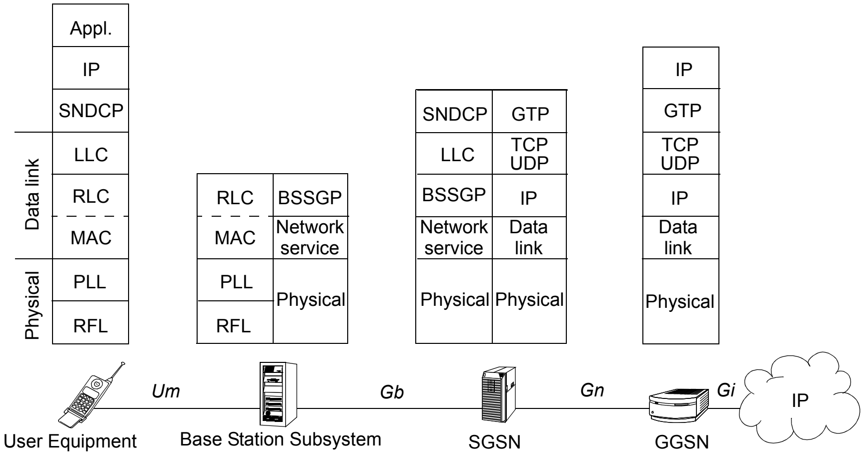


Figure 9.8 The GPRS user plane protocol architecture to connect a mobile user to an IP network. The protocol stack contains two IP layers. One corresponds to the GPRS backbone, the second one corresponds to an IP service network like the Internet

One of the most important protocols within the GPRS network architecture (see Figure 9.8) is the GTP, used in the path between the SGSN and the GGSN (G_n reference point). It employs a tunnel mechanism to transfer user data packets. In the signaling plane, the GTP implements tunnel management and control mechanisms to create, modify and delete tunnels. The GTP can also transport charging information within the GPRS backbone. The GTP is a conventional IP protocol that transports IPv4, IPv6, PPP, X.25 or other packetized user data over a classical TCP/IP network stack. In the lower layers of the protocol stack technologies like Asynchronous Transfer Mode (ATM), Ethernet, Plesiochronous Digital Hierarchy (PDH) or Synchronous Digital Hierarchy (SDH) are used. The GTP exists only between the SGSN and the GGSN. For connections between the user equipment and the SGSN the protocol to be used is known as the SubNetwork Dependent Convergence Protocol (SNDCP). This protocol is compatible with the GSM protocol architecture and provides connection multiplexing and compression of redundant information.

In the radio interface, the protocol architecture has been adapted to support packetized data delivery. The physical layer is made up of two sublayers. The Radio Frequency Layer (RFL) is in charge of modulation/demodulation and the Physical Link Layer (PLL) provides digital transmission channels between the mobile station and the base station subsystem. The data link layer has three stacked sublayers. The Media Access Control (MAC) sublayer provides multiple access to mobile stations, the Radio Link Control (RLC) sublayer provides reliable links over the radio interface by means error recovery mechanisms, and the Logical Link Control (LLC) sublayer, based on a modified version of the LAPDm, provides flow control, packet reordering, and additional error protection.

The interface between the base station subsystem and the SGSN can be built over Frame Relay (FR), ATM or Ethernet. Here, the Base Station Subsystem GPRS application Protocol (BSSGP) is used to deliver QoS and routing information.

Table 9.1
EDGE modulation and coding schemes with their performance

Scheme	*Modulation*	*Speed (kb/s/slot)*
MCS-1	GSMK	8.8
MCS-2	GSMK	11.2
MCS-3	GSMK	14.8
MCS-4	GSMK	17.6
MCS-5	8-PSK	22.4
MCS-6	8-PSK	29.6
MCS-7	8-PSK	44.8
MCS-8	8-PSK	54.4
MCS-9	8-PSK	59.2

Thanks to GPRS, the GSM network is able to efficiently deliver packet services. However, bit rates are still limited to a few tens of kb/s. One of the technologies that attempts to solve this problem is the Enhanced Data rate for Global Evolution (EDGE) or Enhanced GPRS (EGPRS). EDGE uses a new modulation and coding in the radio interface to provide increased data rates. However, the existing core infrastructure is conserved. The modified radio interface can be used both to deliver GPRS and circuit-switched services. Although this technology was first proposed by ETSI as a means to evolving GSM, EDGE is also compatible with IS-136 TDMA systems. This is the reason because some mobile operators, mainly in America, have adopted this technology as the path to 3G for their IS-136 networks.

The basis of the EDGE improvement comes from the 8 Phase Shift Keying (8-PSK) modulation for the radio interface. This modulation offers better performance than the traditional GMSK in environments with low noise and interferences while keeping all other benefits from GMSK (see Table 9.1). EDGE uses 8-PSK in clean environments but it can switch to GMSK if necessary. Both GMSK and 8-PSK can be combined with different channel coding schemes referred as Modulation and Coding Schemes (MCSs). The different MCSs offer specific performance in terms of speed and protection against noise and interference. In multi-slot configurations EDGE supports peak rates over 384 kb/s.

9.1.2 The Universal Mobile Telephone System

UMTS is envisioned as the 3G replacement for GSM. The new system, was initially designed by ETSI but in December 1998, the 3GPP was formed to continue with the technical specification work. Currently, the 3GPP is also responsible for generating standards for the GSM and EDGE systems. Unlike ETSI, The 3GPP is a multiparty organization that involves collaboration of various telecommunications standard bodies known as 3GPP organizational partners: Association of Radio Industries and Businesses (ARIB) from Japan, China Communications Standards Association (CCSA) from China, European Telecommunications Standards Institute (ETSI) from the European Union, Alliance for Telecommunications Industry Solutions (ATIS) from USA, Telecommunications Technology Association (TTA) from South Korea, and Telecommunication Technology Committee (TTC) from Japan.

Table 9.2
Some features added by the 3GPP Releases

Release '99	2000	UTRAN and CDMA air interface.
		ATM based transport within the UTRAN.
		Enhanced Data rates for GSM Evolution (EDGE).
Release 4	2001	Low chip rate TDD air interface (TD-SCDMA).
		Split of the MSC in two functions: MSC server (decision point) and Media Gateway (enforcement point)
Release 5	2002	IP Multimedia Subsystem (IMS).
		IP based transport within the UTRAN.
		High Speed Downlink Packet Access (HSDPA).
Release 6	2005	High Speed Uplink Packet Access (HSUPA).
		Generic Access Network (GAN).
		WLAN Interworking (I-WLAN).
Release 7	2007	High Speed Packet Access evolution (HSPA+).
		Voice Call Continuity (VCC).
		Multiple Input and Multiple Output (MIMO) antennas.

The 3GPP follows a release oriented standardization process (see Table 9.2). New releases of the 3GPP specifications contain new features that improve previous standards or add new functionalities to the UMTS, GSM, and EDGE systems.

UMTS is designed to support both high speed circuit and packet switched services. Theoretical maximum rates supported by UMTS are about 2 Mb/s per mobile user but in real implementations average performance is limited to a few hundreds of kb/s.

9.1.2.1 The WCDMA air interface overview

Unlike GSM users, UMTS users are allowed to share the same frequency even when they are connected to adjacent cells. The reason is that UMTS uses a CDMA-based air interface rather than a FDMA/TDMA interface like GSM. Specifically, UMTS is based in Direct Sequence CDMA (DS-CDMA) technology, that obtains the spread spectrum signal by multiplying the original bit stream by a fast pseudo-random binary sequence (chipping sequence). The original bit stream is recovered in the receiving side by multiplying again by the chipping sequence. In a DS-CDMA system, multiple access is accomplished by assigning chipping codes with low cross-correlation to different users. Orthogonality properties of chipping codes is therefore the basis of multiple access in a CDMA system like it is orthogonality of carrier frequencies in FDMA or time slot orthogonality in TDMA.

There are important reasons to adopt CDMA. Probably the most important of them is the ability of this technology to deal against multipath propagation typical of microwave propagation in mobile applications. Rake receivers for CDMA systems can discriminate multipath arrivals and combine them to make an stronger signal. CDMA signals also exhibit beneficial immunity properties against narrowband interferences. CDMA, however, needs more complex and expensive receivers than other mobile technologies.

The most important UMTS air interface is the WCDMA, that uses different frequency channels for the uplink and the downlink. The WCDMA interface is sometimes referred as the UMTS Frequency Division Duplexing (FDD) for this reason. The alternative to the WDCMA FDD interface is the TD-CDMA that duplexes the uplink and downlink in different time slots within the same radio channel. For this reason, the TD-CDMA interface is known as UMTS Time Division Duplexing (TDD). In fact, TD-SCDMA is also a TDD interface compatible with UMTS and it as been also accepted in the IMT-2000 initiative. The TD-SCDMA technology is a variation of the TD-CDMA developed by China Academy of Telecommunication Technology (CATT) and Siemens operating with slower chipping sequences (1.28 Mcps instead 3.84 Mcps) than TD-CDMA. Generally speaking, TDD is better suited than FDD for data communications due to the increased capability for setting asymmetric transmission channels made up by dynamically assigned time slot groups. On the other hand FDD offers better performance in communications between users moving at high speeds and have less demanding synchronization requirements than TDD.

The ITU-R has allocated spectrum for IMT-2000 / UMTS that has been adapted by national governments in accordance with particular needs and availability leading to a complex scenario. Most of the spectrum used by current rollouts is in the 2 GHz band. Space has been specifically allocated both for FDD and TDD modes, but FDD technology is dominant (see Figure 9.9).

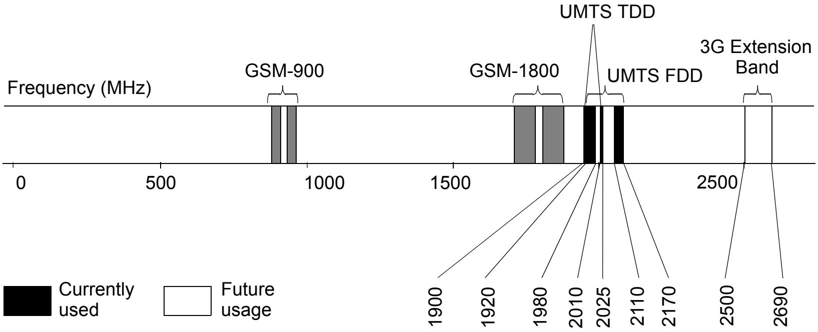


Figure 9.9 Frequency allocations of UMTS in Europe based on ITU allocations. Other 3G systems may operate at slightly different bands

Bandwidth of WCDMA radio channels is about 5 MHz. This is larger that for the case of cdma2000 (1.25 MHz), and much larger than GSM (200 kHz). WCDMA network operators can build their networks with a single carrier frequency for all cells or they can build various overlapped macrocell and microcell structures operating at different frequencies. The standard provides many handoff possibilities:

- *Intra-frequency handoffs* enable mobility between cells operating at the same frequency. This type of handoff is referred as hard handoff if the mobile station is disconnected from the origin base station as soon as it is connected to the destination base station, or soft handoff when the mobile station remains connected to two or more stations at the same time. A special case of soft handoff known as softer handoff, happens when a mobile station is connected to different sectors of the same base station in a sectorial cell made up by directive antennae. Soft and softer handoff do not exist in GSM.
- *Inter-frequency handoffs* enable mobility between stations operating at different frequencies. This kind of handoff is useful to switch between macrocell and microcell layers and makes it possible mobility between different WCDMA operators. Inter-frequency handoffs are always hard.

Handoffs between different systems are also considered by the UMTS standards. Examples are handoffs between WCDMA and TD-CDMA, and handoffs between WCDMA and GSM. Handoffs between WCDMA and non-3GPP networks like Wi-Fi or WiMAX is an active research and standardization area.

WCDMA signals contain 10 ms TDM frames divided in 15 slots. These frames are spread with 3.84 Mcps chipping sequences. There is always 2560 chips in a single time slot but the number of bits per time slot may change. The result is a flexible interface that supports signals with multiple rates. The ratio between the chips and bits in a time slot is referred as the spreading factor. In WDCMA, the spreading factor ranges between 256 and 4 in the uplink, and 512 and 4 in the downlink.

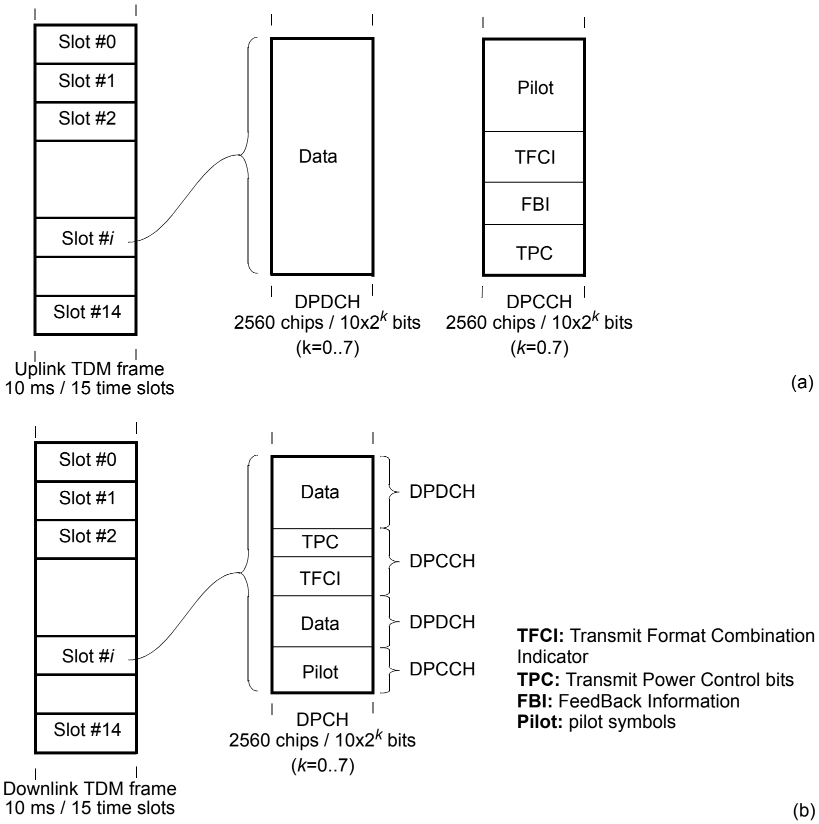


Figure 9.10 Framing of DPDCH, DPCCH and DPCH channels in WCDMA. (a) Uplink frame structure. (b) Downlink frame structure

The WCDMA standard define different types of chipping sequences. Some of them are orthogonal and some of them aren't:

- *Spreading sequences* are used to identify channels. They must be mutually orthogonal if they are transmitted by the same device, otherwise, the receiver will be unable to recover the original information.
- *Scrambling sequences* are used to reduce interference between mobile stations within the same cell and between base stations corresponding to different cells. They are not required to be orthogonal. However, they must have low mutual cross-correlation.

Time slots in WCDMA TDM frames contain traffic and control channels. These channels can be dedicated, shared between a group of users or broadcast channels. The WCDMA standards define a large number of channels with unique features and specific purposes. Depending on the protocol layer being considered by the standard the channel structure is different and channels are identified by different names. As presented by the physical layer, channels are referred as physical channels. Two important physical channels are the Dedicated Physical Data CHannels (DPDCHs) and the Dedicated Physical Control CHannels (DPCCHs). The former contains bearer data, the later contains control and signaling information (see Figure 9.10).

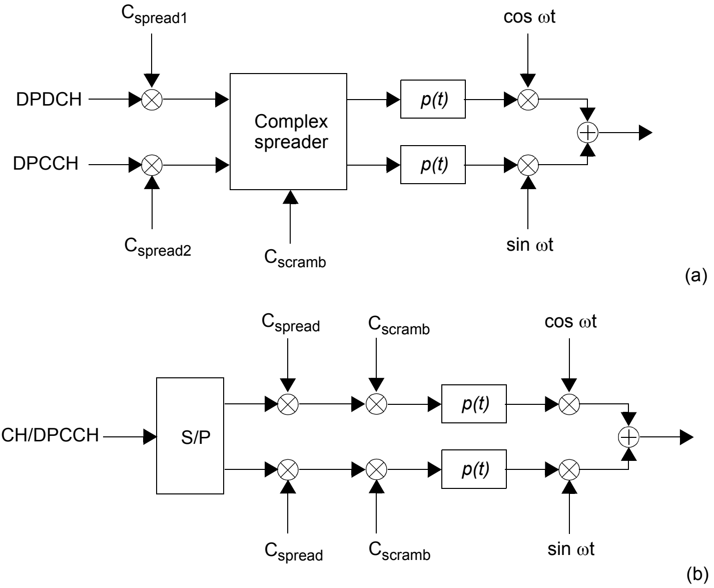


Figure 9.11 Spreading of DPDCH and DPCCH physical channels. (a) Uplink. (b) Downlink.

In the uplink, a DPDCH and a DPCCH (that may have different bit rates) are transmitted in the I and Q phases of a Quaternary Phase Shift Keying (QPSK) modulation after being spread and scrambled. Multiple DPDCHs, spread by unique channelization codes, may be allocated in the same physical layer connection. There is only one DPCCH per connection, however. In the downlink, the information corresponding to the DPDCH and the DPCCH is time multiplexed in a single physical channel known as Dedicated Physical CHannel (DPCH). Chips from the DPCH are modulated in a QPSK signal after being spread and scrambled (see Figure 9.11).

9.1.2.2 High Speed Packet Access

Since its inception in 3GPP Release '99, the WCDMA interface has been improved with important new features. Release 5 specifications defined High Speed Download Packet Access (HSDPA), and Release 6 introduced High Speed Upload Packet Access (HSUPA). Evolved versions of WCDMA are often referred as 3.5G technologies

HSDPA, is a downlink technology that offers increased download bitrate and reduced latency to 3G mobile stations. Specifically, HSDPA increases the UMTS peak bitrate from 2 Mb to 14 Mb/s.

The HSDPA technology relies on a new type of CDMA channel, the High Speed Downlink Shared CHannel (HS-DSCH). Up to 15 of these can operate in a 5 MHz radio channel and user transmissions are dynamically assigned to one or more HS-DSCH channels. The new HS-DSCH is an improvement of the DSCH defined in 3GPP Release '99 that uses various new transmission mechanisms that rely on rapid adaptation of transmission parameters to instantaneous radio conditions. The most important of them are:

- *Fast link adaptation*. Unlike other CDMA channels, the HS-DSCH lacks rapid power control for compensating varying radio conditions. It instead uses rate adaptation. Coding and modulation changes dynamically with time to compensate the time varying radio channel. Spectrum efficiency is improved with the 16-QAM modulation if transmission conditions are good enough. Otherwise 16-QAM is replaced by the QPSK.
- *Hybrid Automatic Repeat reQuest (HARQ).* With this mechanism the system controls transmission errors and requests retransmissions when necessary. This technique combines the advantages of the Forward Error Correction (FEC) and Automatic Repeat reQuest (ARQ) techniques in a single algorithm. HARQ increases system robustness.
- *Channel dependent scheduling*. The base station scheduler considers instantaneous radio channel feedback to determine which is the mobile station best suited to receive data at every moment. This feature improves usage of available transmission resources.

HSUPA uses the new Enhanced Dedicated CHannel (E-DCH) to increase uplink performance of WCDMA networks. Many of the improvements of HSUPA were already known from HSDPA, including fast link adaptation and HARQ. The peak upload bitrate allowed by HSUPA is 5.76 Mb/s. Average performance with realistic transmission conditions is above 1 Mb/s.

HSPA technology is not yet exhausted. Thanks to Multiple Input and Multiple Output (MIMO) antenna systems and other improvements will enable HSPA+ to reach up to 40 Mb/s in the downlink and up to 10 Mb/s in the uplink.

9.1.2.3 The UMTS Network

The UMTS network is an evolution of the GSM/GPRS network and therefore both share many common points. The basis of UMTS is a new radio technology and as a result, main differences with GSM/GPRS are in the radio access network. Two new network elements are introduced, the Node-B is a replacement for the BTS and the Radio Network Controller (RNC) replaces the Base Station Controller (BSC). Together, the Node-B and the RNC make up the UMTS Terrestrial Radio Access Network (UTRAN), that replaces the GSM/GPRS/EDGE base station subsystem. Other GSM/GPRS network elements like the MSC, HLR, SGSN and GGSN) doesn't need to be replaced. However, they have to be extended to adopt the UMTS requirements (see Figure 9.12):

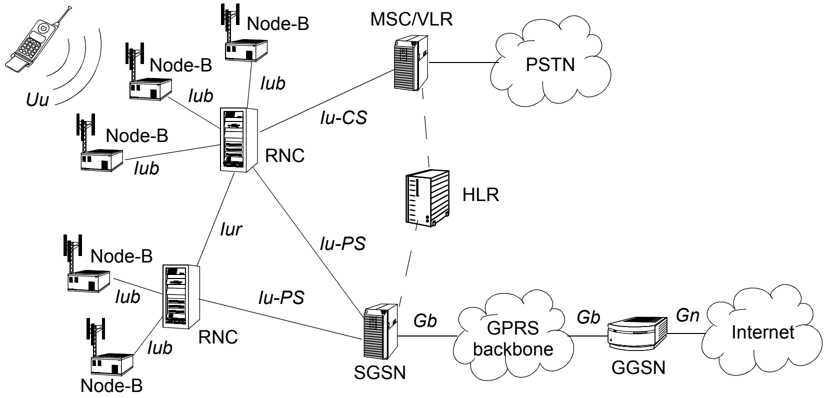


Figure 9.12 UMTS Release '99 network architecture

Release '99 of 3GPP specifications reuses GSM interfaces and define new ones when necessary. The WCDMA interface that connects the user equipment and the Node-B is referred as U_u interface. The UTRAN nodes are connected through the

I_{ub} and the I_{ur} reference points. The I_{ub} connects a Node-B with an RNC and it is similar to the GSM A_{bis} interface. The I_{ur} connects two RNCs and it has no parallel with GSM. The UTRAN is connected to the core network through the I_u interface. Packet-switched services are connected to the SGSN using the I_u-PS interface and circuit-switched services use the I_u-CS to access to the MSC. Both the SGSN and the MSC need to be upgraded to support the new interfaces.

Evolution of the UMTS network architecture is influenced by the development of IP as a new packet-centric transport technology. Releases '99 and 4 define an ATM network for transporting UMTS services but this approach was complemented by an IP based transport alternative defined in Release 5 of 3GPP specifications (see Figure 9.13). The IP transport alternative for the UTRAN is much more flexible than ATM transport. Thanks to the IP transport alternative, mobile network operators can use any third party IP network to interconnect the UTRAN network nodes. QoS control within the IP transport network is one of the issues the network operator needs to consider when migrating to the new architecture. Technologies like MPLS offer a solution to this issue.

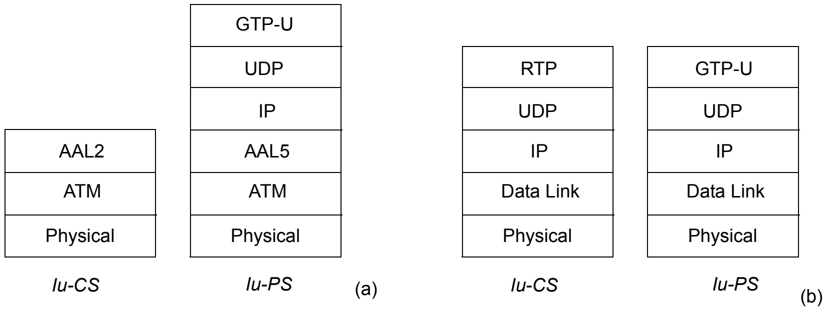


Figure 9.13 Protocol stacks for the different transport alternatives in the Iu interfaces (user plane). (a) ATM transport alternative. (b) IP transport alternative.

IP as influenced the original UMTS network not only as a new transport infrastructure but also in some other ways. The the main of them is perhaps the introduction of the IMS, an advanced multimedia service provision platform based in IP protocols like SIP, SDP and Diameter (see Paragraph 9.3).

9.1.3 Long Term Evolution of 3GPP Networks

HSPA will take the maximum potential to the current WCDMA networks for a few years before being replaced by a new generation of mobile devices. Future 3GPP mobile networks are expected to offer converged communications with the help of single IP core network referred as the All IP Network (AIPN) over an even faster radio access network, the Enhanced UTRAN (E-UTRAN).

Long Term Evolution (LTE) is the name given by the 3GPP to the project that will evolve UMTS to an improved network. LTE will result in the Release 8 of the 3GPP specifications. Initial LTE deployments are targeted for 2009. Some important goals of the project are the following:

- Instantaneous peak data rate of 100 Mb/s (downlink) and 50 Mb/s (uplink) over 20 MHz radio channels.
- Performance targets should be met for 5 km cells with slight degradation for 30 km cells and acceptable performance over 100 km cells.
- Flexible spectrum allocation with radio channels of different bandwidths, including 1.25 MHz, 1.6 MHz, 2.5 MHz, 5 MHz, 20 MHz. FDD and TDD operation will be supported.
- Packet-switched architecture with end to end QoS support.
- Coexistence and interworking with legacy 3GPP systems including UMTS, EDGE and GSM. Handoff between LTE and these architectures should be supported.

The proposed radio interface for the LTE E-UTRAN is referred as High Speed OFDM Packet Access (HSOPA). HSOPA uses Single Carrier FDMA (SC-FDMA) for the downlink and OFDM for the uplink. OFDM was adopted with success by the IEEE WiMAX standard (see Paragraph 9.2.2) but, unlike HSOPA, WiMAX is based in OFDM both for the uplink and the downlink.

OFDM is not compatible with CDMA technologies that dominated 3G, including WCDMA. HSOPA has the same advantages that WCDMA with a simple implementation. Furthermore it offers a more flexible spectrum allocation scheme (spectrum does not need to be allocated in big blocks like it happens with CDMA), and it is well suited for operation with smart antenna technologies like:

- *Beamforming* applies vectorial signal processing techniques to arrays of transmitting or receiving antennas to maximize the transmitted power or sensitivity of the array in chosen directions.
- *Receive diversity* relies in optimal combining of signals received by separate receiving antennas. Antenna spacing yields signals that have somewhat independent fading characteristics and as a result the combined signal may potentially be more effectively decoded that any of the components.
- *Space Time Coding* relies on multiple and redundant copies of a data stream transmitted by multiple antennas. Each copy is encoded using Space Time Codes (STCs). Independent fading in the multiple antenna links makes it possible to decode the received signal.
- *Spatial multiplexing* requires multiple transmitting and receiving antennas in a configuration known as Multiple-Input and Multiple-Output (MIMO). This

technique splits a high rate signal into multiple lower rate streams. Each stream is transmitted from a different antenna in the same radio channel. Signals travelling across different communication paths subject to independent fading can be efficiently recovered.

Although many smart antenna techniques have been specified for WCDMA systems as well, they will become an essential part of the HSOPA interface. In fact, most of the improvement over WCDMA will come from the availability of radio channels with increased bandwidth and smart antenna techniques. HSOPA will possibly use up to four transmitting and up to four receiving antennas.

The access network for the LTE system is an IP network known as E-UTRAN. The E-UTRAN consists in a collection of enhanced Nodes-B (eNBs) connected to an evolved data core. Specifically, eNBs are connected to Mobility Management Entities (MMEs) and Serving GateWays (SGWs) through the *S1* interface. The eNBs can be interconnected as well. The *X2* interface is used for this purpose (see Figure 9.14).

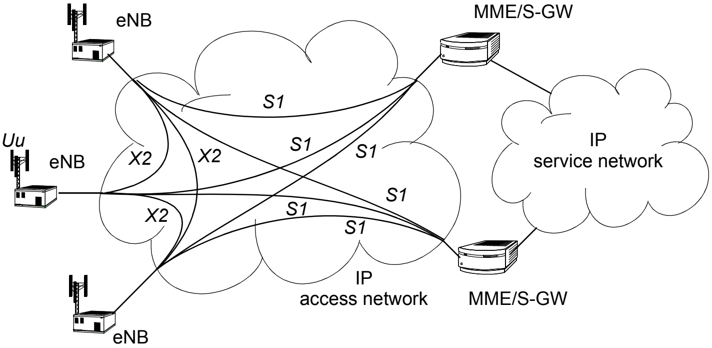


Figure 9.14 E-UTRAN network architecture

The E-UTRAN has a simpler and more efficient architecture than the UTRAN. The new access infrastructure does not have circuit switched domain. All services are provisioned over a single packet switched access architecture connected to a multiservice IP core network: the AIPN. It will be possible for users to access to the AIPN not only by means the E-UTRAN but also by using other technologies like Wi-Fi, WiMAX and wireline.

9.2 Wireless Communications Overview

All communications based on a transmission media other than copper wire, optical fibre, or any other kind of cable, are wireless. This definition includes communications based in satellite microwave links, terrestrial radio links, and free space optical communications. Mobile cellular communications are also included. However, here, the term "wireless" applies only to a few microwave technologies designed for indoor or outdoor access to data networks. This kind of wireless communications is currently dominated by some technologies defined by the IEEE. The most important are the IEEE 802.11 family (Wi-Fi) and the IEEE 802.16 family (WiMAX).

Cellular communications technologies were designed as an extension of the fixed telephone network but evolution has made cellular communications data friendly and suitable for multimedia service delivery. Evolution of IEEE wireless standards has followed the opposite path but destination appears to be the same. The newest IEEE wireless standards are adding sophisticated handoff mechanisms to support mobility, strong QoS for multimedia service delivery and other features that make these technologies suitable for the same applications than WCDMA, cdma2000 or any other mobile standard.

9.2.1 Wireless Local Area Networks

Wireless Local Area Networks (WLANs) provide short range wideband connectivity to computers and other electronic devices. WLAN technologies are designed to operate in indoor environments but they have been used for short range outdoor applications as well. WLAN networks can be either ad-hoc or centralized. In ad-hoc networks, stations are connected through point-to-point wireless links. In centralized networks stations are connected through a central Access Point (AP) rather than by means point-to-point connections.

The applications envisaged for WLANs include:

- *Replacement or complement of wireline Ethernet LANs*. Many businesses have deployed WLAN connectivity in their installations to allow staff access corporate applications or the Internet. WLAN is also used in home networks to connect home computers to the Internet access gateway.
- *Internet access in hotspots*. Hotspots are public wireless networks placed in selected locations like libraries, airports, campuses and train stations. They use to offer Internet access but other services are also possible. Hotspots usually have mechanisms to authenticate, authorize and charge users.
- *Complementary infrastructure for mobile operators*. Current WLAN technologies offer more bandwidth than any mobile access network. Due to its short range, WLAN cannot replace mobile access networks but they can provide

high speed access in selected locations like airports, train stations or the subscriber's home. For this application, mobile operators need dual-mode user equipment with connectivity to mobile and WLAN networks. They must also define the mechanisms to authenticate, authorize and charge subscribers in WLAN mode.

- *Connectivity through mesh networks*. Mesh networks are ad-hoc networks with the special feature that connection with remote nodes can be achieved using intermediate user equipment. Nodes in a mesh network are at the same time terminals and routers. In mesh networks, connection with remote stations is possible even if there are not direct links with them. WLAN technologies can be used to build these networks. Mesh networks may become an alternative wireless access technology to traditional mobile networks.

Currently, the WLAN market is dominated by the IEEE 802.11 family, also known as Wi-Fi. The alternatives to Wi-Fi are, HiperLAN, an European ETSI standard and HiSWAN, a Japanese standard developed by the ARIB. Both with an small market share compared with Wi-Fi.

9.2.1.1 Wi-Fi

Wi-Fi is the branch name of a family of WLAN technologies specified in the IEEE 802.11 standards. One of the advantages of Wi-Fi is that it operates over unlicensed bands of the radio spectrum. This fact eases deployment of Wi-Fi networks. However, interference from external devices is more likely to happen in unlicensed frequencies than in licensed spectrum. Specifically, some Wi-Fi devices operate at the same frequencies than microwave ovens, bluetooth devices and cordless phones.

The first IEEE 802.11 standard was presented in 1997 and today is referred as IEEE 802.11-1997. This standard allows 1 or 2 Mb/s over both infrared and microwave channels. However, infrared operation was never widely accepted by the market. Microwave IEEE 802.11-1997 operates in the 2.4 GHz Industrial, Scientific, and Medical (ISM) radio band. Two spread spectrum modulations were defined for this technology:

- *Frequency Hopping Spread Spectrum (FHSS)*: The spread spectrum signal signal is built by modulating the original bit stream with a narrowband carrier of varying frequency. The carrier frequency must change in accordance with a fast pseudo-random sequence known as frequency hopping sequence. To recover the original signal the receiver must know the frequency hopping sequence and be synchronized with it.
- *Direct Sequence Spread Spectrum (DSSS)*: The spread spectrum signal is obtained by multiplying the original bit stream by a fast pseudo-random sequence (chipping sequence). Binary values '1' and '0' are mapped to '+1' and '-1' dur-

ing the spreading operation. The spreading operation is combined with a Phase Shift Keying (PSK) digital modulation to obtain a pass-band signal. The original signal is recovered by demodulating and multiplying the spread signal by the same chipping sequence again. Pseudorandom sequences must be synchronized in the transmitter and the receiver.

FHSS and DSSS signals have a broader spectrum than the original bit streams. Thanks to this property, signals become protected against narrowband interference. Despite having similar beneficial properties against interferences, it soon became clear that DSSS would offer more possibilities to be scaled to higher speeds than the FHSS an therefore became the preferred modulation.

Wi-Fi did not became really popular until the release of the IEEE 802.11b standard in 1999. IEEE 802.11b devices operate at a maximum speed of 11 Mb/s over the 2.4 GHz band. However, they can switch to 5.5 Mb/s, 2 Mb/s and 1 Mb/s when noise and interferences are high enough to degrade transmission. For 2 Mb/s and 1 Mb/s operation modes, devices are backwards compatible with IEEE 802.11-1997. For 11 Mb/s and 5.5 Mb/s, DSSS is replaced by Complementary Code Keying (CCK). CCK is in fact a variation of DSSS that uses series of special 8-symbol complex sequences called complementary sequences to spread the signal rather than the binary 11-bit Barker sequence used by DSSS. Once modulated, the bandwidth of an IEEE 802.11b signal is about 22 MHz. This signal is placed in one of the fourteen defined radio channels within the 2.4 GHz band[2]. Radio channels are spaced only by an amount of 5 MHz and therefore signal spectra are partially overlapped and interference between neighboring systems is quite likely to occur.

The IEEE 802.11a standard was ratified almost at the same time that IEEE 802.11b. Unlike IEEE 802.11b, the new technology was designed to provide up to 54 Mb/s over the 5 GHz Unlicensed National Information Infrastructure (U-NII) band (see Figure 9.15). However, EEE 802.11a devices are able to modify the transmission rate to 54, 48, 36, 24, 18, 12, 9 and 6 Mb/s depending on the noise and interference level they register. Interferences in the 5 GHz band are less common than in the 2.4 GHz band and thus IEEE 802.11a may potentially provide better performance than IEEE 802.11b.

The IEEE 802.11a modulation is the Orthogonal Frequency Division Multiplexing (OFDM) (see Paragraph 9.2.2.2) that offers excellent protection against narrowband interferences and frequency selective fading caused by multipath propagation. Specifically, IEEE 802.11a OFDM uses 48 data sub-carriers plus 4 pilot sub-carriers. Individual data sub-carriers transport user information using Binary Phase Shift Keying (BPSK), Quaternary Phase Shift Keying (QPSK), 16-Quadrature Amplitude

[2] In some countries only a subset of the fourteen defined radio channels is allowed for 802.11b signals.

Modulation (16-QAM) or 64-QAM. IEEE 802.11a OFDM signals have a bandwidth of 20 MHz. The standard defines 12 non-overlapped channels to carry this signal but only eight of them placed within the lower and mid U-NNI band (5.150 - 5.350 GHz) are currently used.

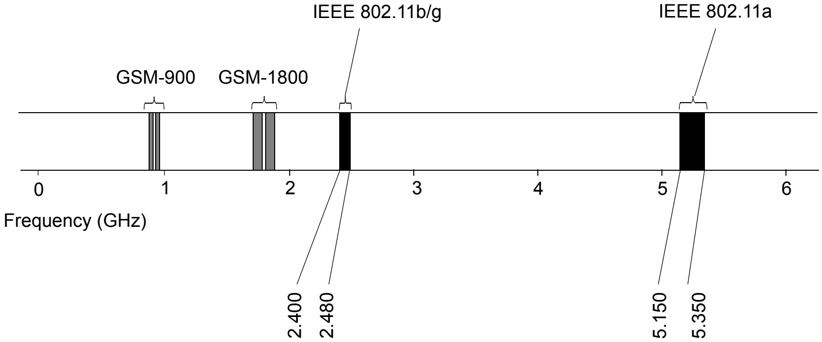


Figure 9.15 Frequency allocations for Wi-Fi.

Although IEEE 802.11a devices provide better performance than IEEE 802.11b, they arrived later to the market at higher price and they were not widely adopted in the consumer space. The primary applications for IEEE 802.11a are found in enterprise network environments. The arrival of IEEE 802.11g in 2003, backwards compatible with IEEE 802.11b and providing similar performance than IEEE 802.11a reduce the chances of mass adoption of the 5 GHz standard. In fact, thanks to its advantages, IEEE 802.11g is currently the dominant WLAN technology in the consumer market.

IEEE 802.11g brings the advantages of OFDM modulation to the 2.4 GHz band. Specifically, OFDM is used for 54, 48, 36, 24, 18, 12, 9 and 6 Mb/s operation modes, CCK for 11 and 5.5 Mb/s, and DSSS for 2 and 1 Mb/s (see Table 9.3). Diverse modulations for different bit rates makes IEEE 802.11g compatible with older devices. Specifically, this feature makes IEEE 802.11b and 802.11g interoperable although the presence of a IEEE 802.11b participant reduces the speed of the overall IEEE 802.11g network.

Compatibility between IEEE 802.11a and 802.11g devices is more difficult as they operate in different bands. However, there exist Wi-Fi access points with dual band capabilities. These devices can operate both at 2.4 GHz an at 5 GHz bands.

Table 9.3
Comparison between current Wi-Fi technologies

Standard	*Release Date*	*Operation band*	*Modulation*	*Throughput (typ/max)*	*Range (indoor)*
IEEE 802.11a	October 1999	5 GHz	OFDM	23/54 Mb/s	30 m
IEEE 802.11b	October 1999	2.4 GHz	DSSS/CCK	5/11 Mb/s	35 m
IEEE 802.11g	June 2003	2.4 GHz	OFDM/DSSS/CKK	19/54 Mb/s	35 m

Despite the many different modulations used by different versions of the IEEE 802.11 family, all them rely in the same MAC protocol known as Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) that is a variation of the CSMA with Collision Detection (CSMA/CD) used by IEEE 802.3 wired Ethernet. Radio devices are unable to detect collisions, therefore, they use collision avoidance based on a listen before talk and random back off deferral mechanism. The contention-based multiple access mechanism of Wi-Fi is simple and does not need centralized control. Performance is good enough for most data applications and ideal for easy deployment of mesh networks. However, applications like wireless VoIP may need more predictable MAC protocol.

9.2.1.2 The future of Wi-Fi

The market is waiting for a new generation of Wi-Fi devices driven by new applications like wireless VoIP, streaming video and music, gaming, and network storage. Next release of IEEE 802.11n is expected to increase average bitrate in the radio interface to at least 100 Mb/s. IEEE 802.11n is still waiting for a final approval but some vendors have started manufacturing using draft version specifications.

IEEE 802.1n is expected to be compatible with older Wi-Fi standards and at the same time it will offer important improvements. Perhaps the most important of them will be support of Multiple-Input and Multiple-Output (MIMO) antenna systems. These systems use multiple transmitting and receiving antennas to improve transmission performance. Other improvement will probably come from usage of 40 MHz ODFM signals rather than operation limited to 20 MHz signals like in IEEE 802.11g.

9.2.1.3 WLAN integration in mobile networks

The telecommunications industry has been searching for solutions to merge the high availability of cellular networks with the high speed and low cost of WLANs. Dual-mode user equipment remain connected to the cellular network like other mobile device but they can switch to WLAN operation when there is a WLAN available for access. 2G/3G cellular voice services are provisioned through the circuit switched domain of mobile core network while WLAN services are commonly offered through a packet switched, fixed, IP-centric infrastructure. This is the reason because convergence between cellular and WLAN communications is considered a key step towards Fixed Mobile Convergence (FMC). The requirements to any solution aiming to bring WLAN access to mobile services are:

- *Service transparency.* The services available to dual-mode terminals must be the same in cellular and in WLAN operation or, at least, the common set of services available must be large enough to be attractive to potential subscribers. Uninterrupted service provision by means seamless handoff between WLAN and mobile networks is important as well.
- *Seamless integration to current mobile network.* The convergence solution should cause minimum disruption in available network infrastructure. It should be possible to reuse most or all the current network nodes and the solution should not disturb services already deployed.
- *Scalability to support mass market deployments.* A successful WLAN/mobile convergence solution should be scalable enough to address the residential market. Fast rollout, availability of cheap terminals and simple service configuration and operation are a must.

So far, two main approaches for mobile/WLAN converged services have been considered. Both with some advantages and some inconveniences:

1. *Solutions based in specialized access gateways.* These approaches are based in a new network element that translates WLAN signals and protocols to the mobile network. Management of WLAN terminals connected to the mobile core network through this new gateway is done like if they were ordinary mobile terminals.
2. *Solutions based in IMS.* IMS is a platform that enables delivery of multimedia services through heterogeneous access networks (see Paragraph 9.3). It is therefore normal to think that IMS should enable seamless handoff between different access networks for defined multimedia services.

The most important initiative based in access gateways is the Generic Access Network (GAN) also known as Unlicensed Mobile Access (UMA). The IMS-based solution is the referred as Voice Call Continuity (VCC) (see Paragraph 9.3.2). GAN is a 3GPP standard that connects WLAN users to the G_b and A interfaces of the GSM/GPRS/EDGE network with the help of a gateway known as GAN Controller (GANC) (see Figure 9.16). Current GAN standard only provides IP access to the GSM G_b and A reference points and therefore, interworking with UMTS networks and services is limited. On the other hand, VCC, also being defined by the 3GPP, is a core network technology independent of the access network. Theoretically, it should work with Wi-Fi, WiMAX, WCDMA, wireline, and any other access network but it is limited to the voice service. It could be necessary to define more IMS continuity applications for other services in the future. In must me remarked that IMS and GAN may work together as well. Specifically, an operator may choose to deliver IMS multimedia services to WLAN and mobile access points through a GAN access infrastructure. In this case VCC is not needed.

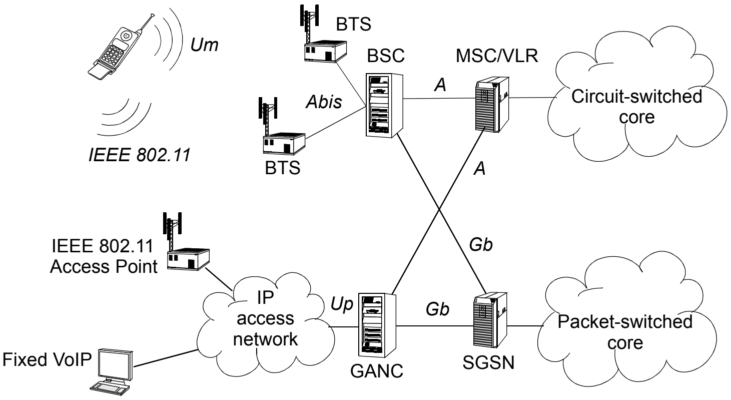


Figure 9.16 UMA offers seamless integration of WLAN access points and the GPRS/EDGE radio access network thanks to a new network element known as GANC

An alternative access network to GAN is the architecture known as WLAN Interworking (I-WLAN) (see Paragraph 9.2.2.3).

The basis of the GAN standard is the new U_p interface that maps GSM/GPRS/EDGE protocols to IP datagrams. The IP traffic is collected by the GANC and translated to G_b and A signals and protocols that can be interpreted by the GSM/GPRS/EDGE core network. For this reason, the GANC appears to the core network as a normal BSC and this enables mobile operators to use any IP access network for provision of mobile services. Usually, mobile operators attach to the IP access networks WLAN access points, but this does not need to be the case. For example, a DSL operator may use GAN to provide fixed VoIP services to their subscribers through a mobile network. This approach would need special VoIP phones with GAN support.

Any IP network, including the Internet, can be theoretically used by GAN as the access infrastructure. QoS capabilities of this network are important for high quality voice service provision. However, QoS is always limited in Wi-Fi access networks by the own nature of the IEEE 802.11 interface. QoS may be further degraded if the access infrastructure is the Internet due to the lack of QoS mechanisms of this network.

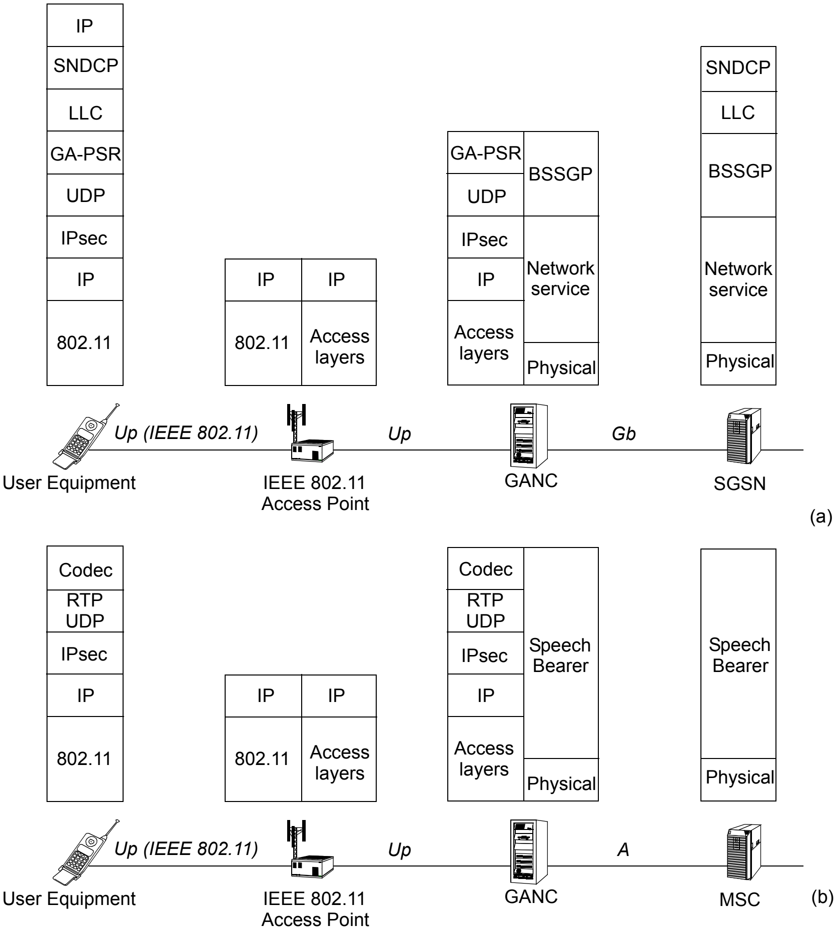


Figure 9.17 GAN user plane protocol architecture. (a) IEEE 802.11 access to the GPRS packet-switched domain. (b) IEEE 802.11 access to the GSM circuit switched domain. Voice is encoded, packetized and delivered through the IP access network. Timing information is transported with the help of the RTP protocol.

Another issue with public IP access infrastructure is security. The GAN standard makes it use of IPsec Encryption Security Envelope (ESP) tunnels to provide encryption and data integrity to voice, data and signaling traffic (see Figure 9.17). IPsec tunnels are established as a result of authentication and authorization procedure of the identity carried by a smart card located in the user equipment. Authentication and authorization is performed by a SEcurity GateWay (SEGW) included in the GANC. GAN authorization and authentication is based on standard IETF and 3GPP procedures. Additional security may exist in GAN networks. For example, the user needs to be authenticated by the mobile core network before accessing to mobile services. This authentication procedure happens between the user equipment and the MSC/VLR. GPRS services also have their own security mechanisms and some applications like the web or E-mail may need end-to-end specialized security as well.

User data and signaling are delivered over previously established IPsec tunnels between the user equipment and the GANC. GSM/GPRS/EDGE protocols are used when possible and specific GAN protocols are defined when necessary. Voice is transported in the IP network with the help of the Real-time Transport Protocol (RTP) over UDP. RTP/UDP allows delivery and recovery of timing information carried by isochronous signals. The GPRS protocol stack is conserved as much as possible. Specifically, the LLC and SNDCP protocols does not need to be modified. GPRS data is transported over UDP. Timing recovery is not necessary for data applications and therefore RTP is not used. Control plane signaling is sent over reliable TCP connections

There are three GAN-specific protocols for the U_p interface. The Generic Access - Circuit Switched Resources (GA-CSR) is used in circuit switched services, the Generic Access - Packet Switched Resources (GA-PSR) is used in packed switched services and the Generic Access - Resource Control (GA-RC) is used for both circuit and packet switched services.

- *GA-RC*: It is a control plane protocol connects users to the GANC. It provides discovery and registration mechanisms for mobile stations.
- *GA-CSR*: Control plane protocol that provides resource control mechanisms to GAN equivalent to the GSM RR protocol. It establishes and maintains communication paths for circuit-switched services and provides handoff coordination of circuit switched services between the GSM/GPRS/EDGE and GAN networks.
- *GA-PSR*: It is a user and control plane protocol that transports GPRS signaling, the SMS service and user data. It manages GPRS transport channels over the U_p and provides handoff coordination between either the GSM/GPRS/EDGE and GAN networks or the UTRAN and GAN networks.

9.2.2 Wireless Metropolitan Area Networks

Wireless Metropolitan Area Networks (WMANs) technologies were designed to be a first mile wireless alternative to cabled networks such as copper based Digital Subscriber Loop (DSL), optical fibre Passive Optical Networks (PON) or coaxial-based Data Over Cable Service Interface Specifications (DOCSIS) access technologies. A secondary application envisaged for WMAN is low cost backhauling for wireline, WLAN and cellular networks.

Most of the current market interest in WMAN is currently focused in IEEE 802.16 family of standards, commonly referred as WiMAX. The WiMAX standardization activity is responsibility of IEEE Working Group 802.16 on Broadband Wireless Access Standards that started its activities in 1999. Other organization involved in promoting WiMAX is the WiMAX Forum. The WiMAX Forum was formed in 2001 by operators and equipment vendors with the objective of ensuring compatibility and interoperability of WiMAX communication equipment.

The first WiMAX standard was the IEEE 802.16 released is January 2001, commonly referred as IEEE 802.16-2001. This standard included a physical and MAC layer specification for WiMAX operating in the 10-66 GHz band using a single carrier modulation. This band is well suited to deliver high data rates for point-to-point, line-of-sight applications. Backhauling for access networks is a good application for the 10-66 GHz band but this band it is not optimized for point-to-multipoint, non-line-of-sight applications such as broadband wireless access. The standard IEEE 802.16a-2003, fixes this issue specifying operation in the 2-11 GHz band, much better suited for broadband wireless access than the 10-66 GHz band. The currently active base WiMAX standard is the IEEE 802.16-2004 sometimes referred as IEEE 802.16d. However, the latest important improvement about WiMAX so far are included in the standard IEEE 802.16e-2005 that includes handoff procedures for IEEE 802.16 networks thus enabling terminal mobility. Thanks to IEEE 802.16e-2005, WiMAX becomes an alternative to 3GPP cellular networks. Mobile WiMAX probably will be an important ingredient of the future mobile network architecture.

Despite being both IEEE standards, WiMAX and Wi-Fi are different technologies. That means that WiMAX and Wi-Fi user and network equipment are not interoperable. The reason is that Wi-Fi and WiMAX have been designed for different applications and therefore they have different technological requirements.

The performance of WiMAX technology in terms of range and bit rate is difficult to state because it depends largely on system configuration. Specifically, performance depend on the antennas, radio power, operation band, radio channel bandwidth and many other parameters. It has been reported that WiMAX operating in the 3.5 GHz band may provide a 10 Mb/s downlink over 10 km line-of-sight paths in a pair of 3.5 MHz radio channels. However, more realistic results for non-line-of-sight paths point out to 10 Mb/s downlinks in 2 km paths. Performance tends to be better in rural and suburban areas than in urban environments. The given 10 Mb/s bit rate are aggregated capacity that needs to be shared among all users of the same cell and radio channel.

Main alternatives to WiMAX are WiBro, a Korean standard, and HiperMAN, developed by ETSI. Both are very close to WiMAX and they are interoperable with this technology or at least with some of its profiles.

9.2.2.1 Allocated radio bands for WiMAX

Allocation of frequency bands for WiMAX is more difficult than for Wi-Fi. Wi-Fi exists in unlicensed spectrum, vendors can manufacture and sell products without heavy constraints. However, broadband wireless access business models demand licensed frequencies subject to a close control by governments. Reaching international agreement on spectrum issues is complicated and is an issue still unsolved for WiMAX. This may delay or limit the adoption of WiMAX and affect the interoperability of network equipment.

There are, however, various licensed and unlicensed frequency bands that are the most likely to be used for broadband wireless access based on WiMAX.

- The 3.5 GHz band is the most widely available band outside the US, including many European and Asian countries. This band contains licensed spectrum and it represents the largest potential market for WiMAX.
- The 5 GHz Unlicensed National Information Infrastructure (U-NII) and World Radio Conference (WRC) bands, represents other major spectrum slice available for WiMAX. In this case they contain unlicensed frequencies. The low and mid U-NNI are already being used by IEEE 802.11a Wi-Fi systems. Most WiMAX activities are in the upper U-NNI band because it contains fewer interferences.
- The Wireless Communications Service (WCS) bands are two narrow spectrum bands around 2.3 GHz that can be used in the US for licensed applications.
- The Multichannel Multipoint Distribution Service (MMDS) band is available in the US for licensed applications. This band contains 31 channels with 6 MHz spacing originally allocated for the Instructional Television Fixed Service (ITFS) and due to its under-utilization has been allocated for broadband wireless access applications.
- The 2.4 GHz Industrial, Scientific, and Medical (ISM) radio band has been used for IEEE 802.11b/g Wi-Fi applications. And it is also available for WiMAX broadband wireless access.

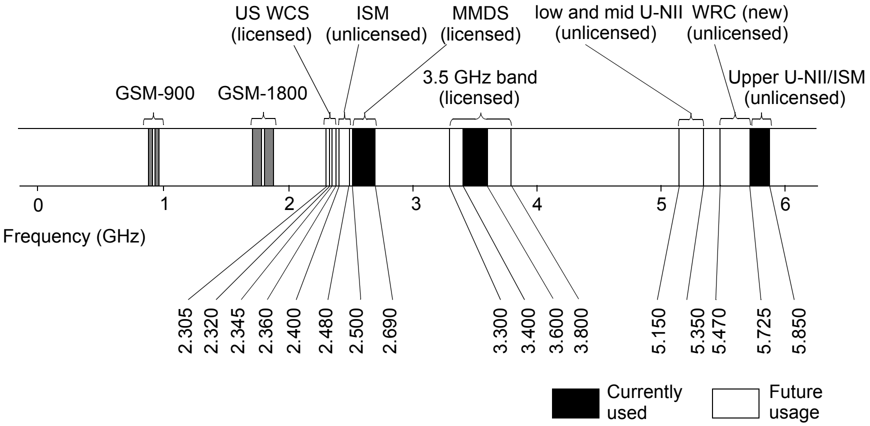


Figure 9.18 Frequency allocations for WiMAX

9.2.2.2 WiMAX air interface specification summary

For the physical layer, WiMAX uses different versions of Orthogonal Frequency Division Multiplexing (OFDM). A single carrier modulation from the first IEEE 802.16 release is still included in the standard but not implemented in commercial devices. None of the physical layer specifications for WiMAX supports CDMA. The OFDM waveform is composed by many narrowband orthogonal carriers. In a OFDM system, the input data stream is divided into several parallel sub-streams of reduced data rate and increased symbol duration. Each sub-stream is then transmitted on a separate orthogonal sub-carrier using single carrier digital modulations such as the Binary Phase Shift Keying (BPSK), Quaternary Phase Shift Keying (QPSK), 16-Quadrature Amplitude Modulation (16-QAM) or 64-QAM.

Increased symbol duration of OFDM provides good protection against Inter-Symbol Interference (ISI) and delay spread typical of non-line-of-sight radio communications. The multi-carrier modulation is beneficial against frequency selective fading caused by multipath propagation because attenuation of some sub-carriers does not affect the others. OFDM radio systems have demonstrated to be a good way to fight against multipath propagation with less complexity than CDMA systems. Moreover, OFDM systems can achieve higher spectral efficiency than CDMA systems.

Strictly speaking, OFDM does not provide multiple access in the same sense that CDMA does. To accommodate multiple users, OFDM relies either in Time Division Multiple Access (TDMA) or Frequency Division Multiple Access (FDMA) mechanisms. Specifically, in an OFDM system, transmission resources are available in the time domain by means OFDM symbols and in the frequency domain by means of sub-carriers. Both sub-carriers and OFDM symbols are distributed among users in order to provide multiple access.

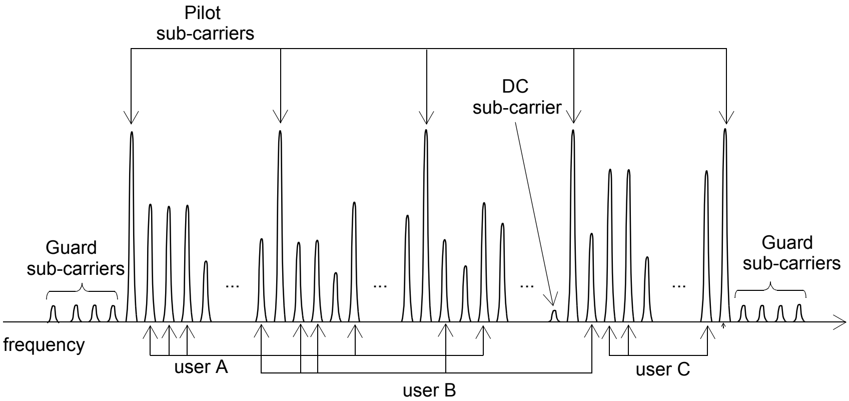


Figure 9.19 OFDMA operation principle

Orthogonal Frequency Division Multiple Access (OFDMA) is a variation of the OFDM modulation that achieves multiple access by assigning subsets of sub-carriers, known as sub-channels, to individual receivers (see Figure 9.19). Sub-channels can be resized or reassigned to different sub-carriers with the help of an adaptive algorithm and using the feedback information from the channel conditions. Differentiated service levels can be achieved by assigning different number of sub-carriers to different users. The version of OFDMA for WiMAX is referred as Scalable OFDMA (S-OFMDA) because it fixes frequency spacing to 10.94 kHz for radio channels of different bandwidth. This can be done by modifying the total number of sub-carriers in radio channels of different size (see Table 9.4). Thanks to this feature, S-OFDMA is able to support a wide range of bandwidths to address the need for various spectrum allocation and usage model requirements.

Table 9.4
OFDMA sub-carriers and frequency spacing for different radio channels in mobile WiMAX

Parameter	*Transmission profiles*			
Radio channel bandwidth (MHz)	1.25	5	10	20
Number of sub-carriers	128	512	1024	2048
Sub-carrier Frequency Spacing (kHz)	10.94	10.94	10.94	10.94

S-OFDMA was introduced in the IEEE 802.16e-2005 standard for mobile WiMAX. Before the release of this standard, the WiMAX Forum established a transmission profile based in OFDM modulation with 256 sub-carriers for fixed services. The profiles that are most likely to be adopted by the market for mobile WiMAX are the S-OFDMA with 512 and 1024 sub-carriers. The latter, matches Korea's WiBro specification.

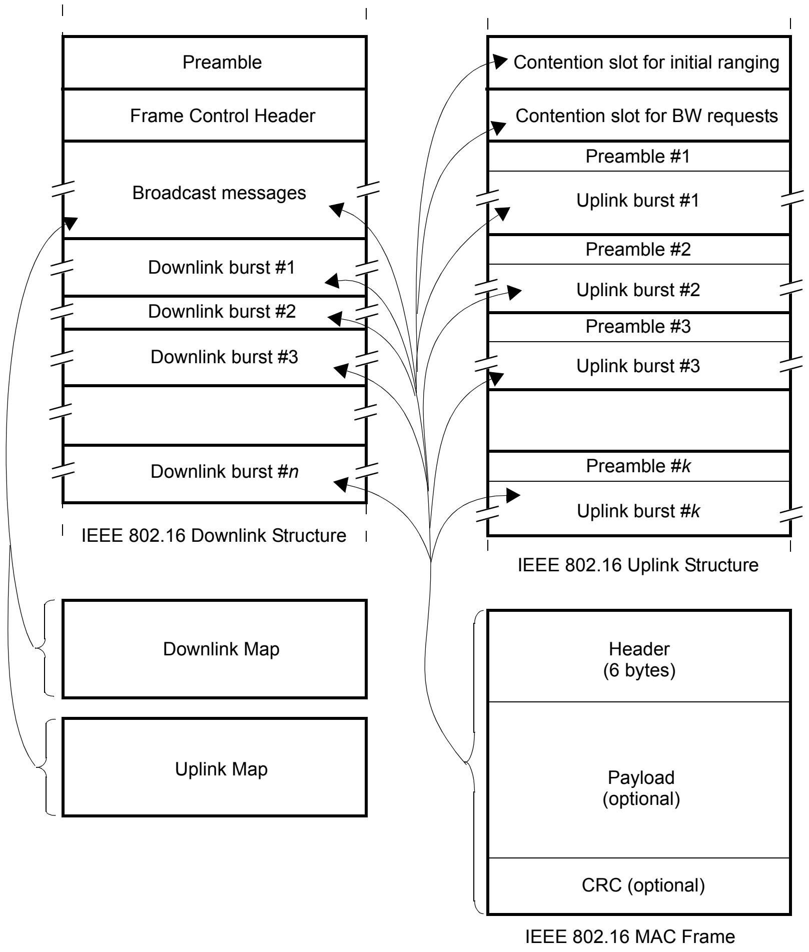


Figure 9.20 Simplified structure of the uplink and the downlink in IEEE 802.16 WiMAX. The downstream uses TDM and mobile stations share the radio channel using TDMA. MAC frames from different users are inserted in TDM bursts associated to different mobile stations. Uplink TDMA bursts are preceded by preambles. The uplink and downlink maps provide information about how time is distributed between users. Some uplink bursts are shared between some or all mobile stations. In these bursts collisions may happen.

WiMAX supports both TDD and FDD duplexing of the uplink and downlink. In TDD mode, uplink and downlink bit streams share a single frequency but are transmitted at different times. In FDD mode, two different frequencies are allocated for the uplink and the downlink. There is also an half duplex FDD mode that only needs one single frequency.

The WiMAX MAC is designed to support point-to-multipoint communications with uplink and downlink QoS handling that enable concurrent and reliable transmission of bandwidth demanding applications such as encoded video, bursty traffic such as data and delay sensitive traffic such as voice. To meet these requirements WiMAX uses a connection-oriented MAC layer based on flexible TDMA to allocate bandwidth for different user equipment.

The WiMAX MAC layer has a service specific, and a common part sublayers. The service specific sublayer defines encapsulations for cell services (ATM) and packet services (IP, Ethernet). The common part sublayer addresses management of multiple access over a shared same radio channel.

All traffic between base stations and mobile stations is mapped into connections identified with a Connection IDentifier (CID). Base and mobile stations use the CID to short and schedule MAC frames. Connections are therefore the basis of the QoS provision in the air interface. Base stations deliver traffic to mobile stations attached to them using TDM frames. The uplink is based on TDMA (see Figure 9.20). Mobile stations request uplink bandwidth to the base station and the base station grants bandwidth to mobile stations. The base station broadcasts information about how downlink and uplink transmission time is distributed between mobile stations in the DownLink map (DL-map) and the UpLink map (UL-map). The process by how base stations allocate bandwidth to the mobile stations for the specific purpose of making bandwidth requests is called polling. There are three different polling modes. QoS in the air interface strongly depends on the selected polling mode:

1. *Unicast polling* happens when mobile stations send bandwidth request messages in uplink bursts allocated for their exclusive use. No explicit polling message from the base station is necessary. The base station simply allocates bandwidth in the UL-map to allow the mobile station to send bandwidth request messages in its time slot.
2. *Multicast/broadcast* polling is possible if the bandwidth allocated in the UL-MAP is associated to a multicast or a broadcast CID rather than to a unicast CID. The mobile station is then allowed to issue bandwidth request messages in the contention slot allocated for that CID. Collisions in multicast/broadcast bandwidth requests may happen in contention slots. This means that bandwidth grants are subject to unpredictable delay when multicast/broadcast polling is used and therefore this operation mode is not

recommended for real-time applications

3. *Unsolicited grant* does not need explicit bandwidth request messages. A fixed amount of bandwidth is reserved in periodic time intervals known for the mobile station.

All uplink connections are mapped to one of the five possible uplink scheduling services. Every uplink scheduling service is suited for a specific family of applications. The defined scheduling services are:

- *Unsolicited Grant Service (UGS)* is best suited for real time services like PCM voice or emulation of circuit-switched services. UGS services does never use contention-based bandwidth request mechanisms to ensure maximum predictablity.
- *real-time Polling Service (rtPS)* is designed to carry real time service flows that generate variable size data packets on a periodic basis such as MPEG video. These services use unicast polling.
- *Extended rtPS (ErtPS*) is a mixture between the UGS and the rtPS scheduling services. ErtPS relies on unsolicited unicast grants like the UGS but carry variable size data packets like the rtPS.
- *non-real-time Polling Service (nrtPS)* is suited to carry service flows that generate variable size data packets. Unlike rtPS, real-time operation is not required for these services. A typical example of nrtPS is FTP.
- *Best Effort (BE)* service offers efficient transport of best effort traffic such as Internet data traffic. Contention-based bandwidth requests are allowed. Unicast polling and unsolicited grant operation is possible as well.

Other features supported by the WiMAX physical layer include different handoff modes, traffic encryption, and key management. However, the most important WiMAX advanced feature is the availability of a full range of smart antenna technologies which enormously increase system range and capacity

9.2.2.3 WiMAX and 3GPP network integration

WiMAX is called to play an important role in future mobile infrastructure. This is the reason because interworking between WiMAX and 3GPP networks is becoming an important topic. The same requirements and the same solutions apply both to WLAN and WMAN integration with 3GPP networks (see Paragraph 9.2.1.3). That means that GAN, VCC and IMS are suitable solutions for different levels of integration between with 3GPP network.

There is a critical difference, however, between Wi-Fi/3GPP integration and WiMAX/3GPP integration. Unlike Wi-Fi, WiMAX aims to become a full alternative to 3GPP networks. Pointing towards this direction, the WiMAX Forum has specified a reference model for mobile networks based in the IEEE 802.16 air interface that includes all the network elements, interfaces, and procedures to operate as an standalone network. This does not happen with Wi-Fi, with reach usually limited to a few tens of meters. While integration between Wi-Fi and 3GPP networks is a problem of extending the original 3GPP infrastructure to a new air interface, integration between WiMAX and 3GPP resembles more a problem of interworking between two heterogeneous networks.

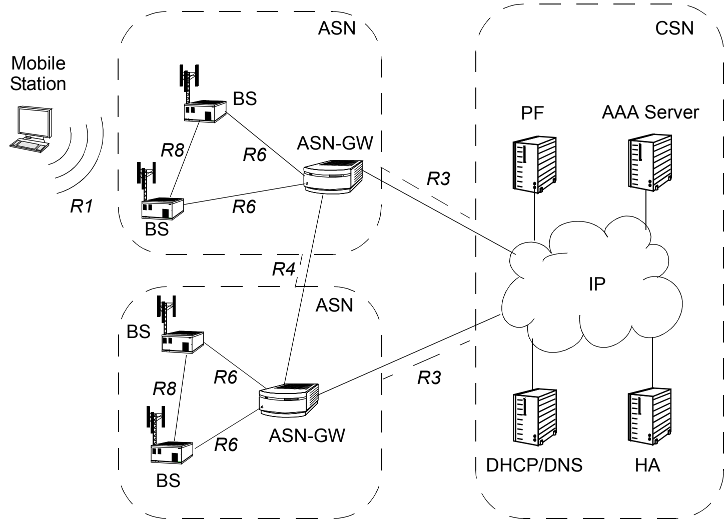


Figure 9.21 Simplified WiMAX network reference model as defined by the WiMAX Forum.

The WiMAX Forum network reference model is based in IETF and IEEE signals and protocols. This reference model is very flexible and enables network operators to choose between different physical configurations. The WiMAX network is made up by an Access Service Network (ASN) and a Connectivity Service Network (CSN) connected through an ASN GateWay (ASN-GW) (see Figure 9.21). The WiMAX Forum reference model defines several reference points (*R1*, *R2*,...) that enable communication between the different entities that made up the network. Every reference point has its own associated user plane and control plane protocols and procedures. IETF and IEEE protocols are reused when possible. WiMAX Forum network includes procedures for network discovery, IP address management, Authentication, Authorization, and Accounting (AAA), QoS provision, CSN and ASN anchored handoffs, etc.

The WiMAX Forum has defined an interworking model between WiMAX and 3GPP networks based in the 3GPP I-WLAN standard. I-WLAN is a 3GPP standard defined in TS23.234 and TS33.234 that provides several interworking levels between WLAN and 3GPP mobile networks. In the simplest scenario, I-WLAN allows authentication and billing of wireless user equipment through the 3GPP mobile network. This has been also used for Internet access in public Wi-Fi hotspots managed by mobile operators. A more complex deployment scenario for I-WLAN also allows access to packet-switched services provided by the mobile network.

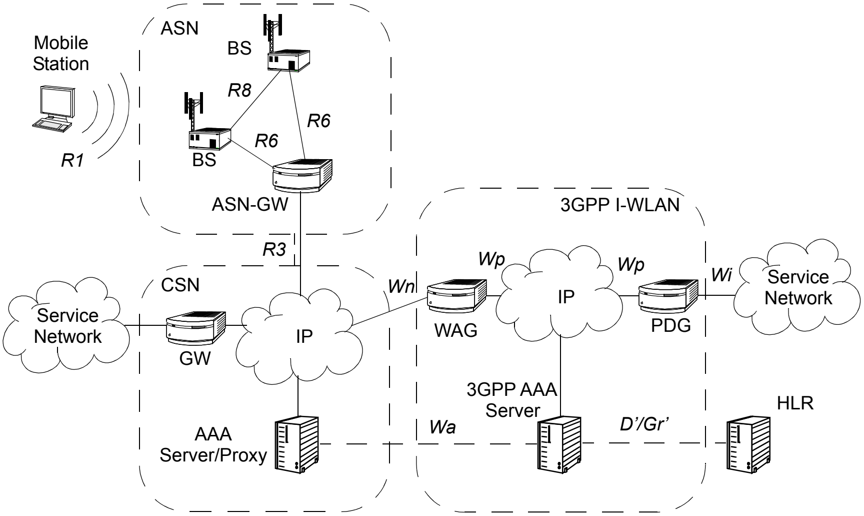


Figure 9.22 Simplified WiMAX and 3GPP interworking architecture based in the I-WLAN standard.

I-WLAN allows direct interconnection of the WiMAX and 3GPP core network without defining new interfaces or protocols. In fact, the WiMAX network is more complex that WLAN networks considered by the I-WLAN standard an therefore this standard needs to be customized for this particular application. The WiMAX Forum builds the WiMAX/3GPP interworking solution on top of the I-WLAN specification without modifying it.

Most interconnection issues are related with AAA. The interworking solution adds an AAA Server/Proxy in the AAA signaling path between the user equipment and the I-WLAN AAA Server (see Figure 9.22). The I-WLAN AAA Server is the entity that authenticates users and helps with any other AAA procedure over the WiMAX network and users connected to it. However AAA signaling messages always traverses the AAA Server/Proxy placed in the WiMAX CSN. The I-WLAN AAA Server may also need to retrieve user profile information from the HLR or other 3GPP databases. For this reason, the I-WLAN standard defines interfaces with these databases.

Integration between WiMAX and the 3GPP network may be limited to unified registration and billing of may also allow access to packet-switched services through the 3GPP network. If this is the case, the WiMAX CSN is connected to the I-WLAN WLAN Access Gateway (WAG) through the W_n reference point. Information traversing the WAG is tunneled to an I-WLAN Packet Data Gateway (PDG) using the W_p interface before it enters in the network that implements the services required by WiMAX users.

Work on integration of WiMAX and other networks is just starting. While the I-WLAN architecture is a good starting point, more integrated interworking between WiMAX, 3GPP and other networks are expected to appear in the future.

9.3 The IP Multimedia Subsystem

The Public Switched Telephone Network (PSTN) plays a central role in the business value chain associated to the Plain Old Telephone Service (POTS). However, things are less clear if the PSTN is replaced by the Internet and the POTS is replaced by IP telephony. In fact, many companies have succeed in building viable business models around the Internet without consent of any network operator. IP telephony service providers are just one of the many possible examples of successful business model built over the Internet. Others are search engines, online shops, network games and document repositories. Moreover, Peer-to-Peer (P2P) applications are becoming increasingly important in the Internet but network operators fail to control or manage these applications with the currently available mechanisms at their disposal.

It can be said that IP and the Internet have the effect of moving the value of telecommunications services from the network to the end equipment and thus, IP make it difficult for network operators to obtain high return to their investments. Network operators are facing the risk of becoming mere “bit pipe” administrators. Today, telcos are seeing how their network becomes a commodity but they are unable to find an alternative to IP converged networks due to two reasons:

1. They can not ignore the success of IP services. Being in the IP market is a must for them.
2. They must keep operating expenditure and capital expenditure as low as possible to be competitive. Today this can only be made through a single multiservice IP network.

The solution that the telecommunications industry has proposed to keep the value of IP-centric network operators is the IP Multimedia Subsystem (IMS). In other words, IMS is a way to keep applications user-centric in a network-operator-centric business model.

IMS is a session control subsystem based on IP, SIP, SDP, and other protocols, specifically designed for provision of multimedia services through high variety of access networks (see Figure 9.23). As well as session control, IMS provides subscriber profile management, charging mechanisms and bearer and QoS resource allocation for media transmission.

IMS was introduced in 3GPP specifications Release 5 in June 2002. The initial IMS specification was enhanced in Releases 6 and 7 and it is still an active standardization area. IMS brings together the IETF and the 3GPP worlds. Currently, the organization that releases IMS standards is the 3GPP but most of the IMS protocols are largely based on IETF RFCs.

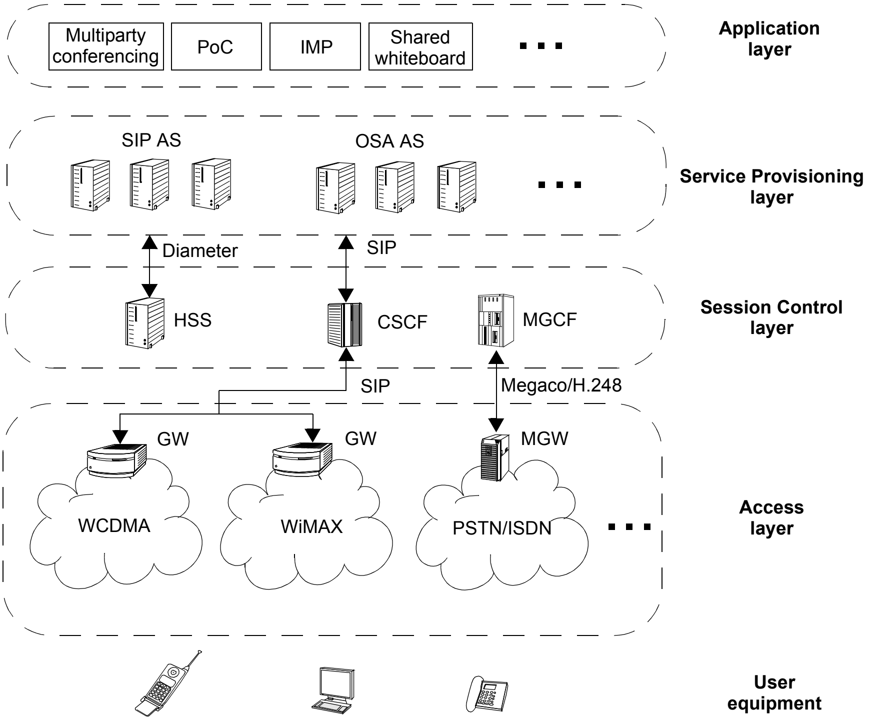


Figure 9.23 IMS layered architecture

The key components of the IMS architecture are the Call Session Control Function (CSCF) and the Home Subscriber Server (HSS). Together, these components control how users access to the services in a highly versatile and customizable environment. The power of IMS is manifested through the great variety of user terminals that potentially could log on in the IMS capable network through many types of access mechanisms. Supported user equipment include mobile phones but also Personal Digital Assistants (PDAs), computers, VoIP phones, gaming consoles and many others. These devices must support the IPv6 protocol stack and run SIP user agents. Supported access networks are:

- *Wireline networks* based on Digital Subscriber Loop (DSL), Passive Optical Network (PON), Data Over Cable Service Interface Specifications (DOCSIS) or Ethernet.
- *Cellular mobile networks* including WCDMA, cdma2000, GSM and GPRS
- *IEEE wireless networks* including Wi-Fi and WiMAX.

Applications potentially available for IMS subscribers is large as well. Some example applications are video telephony, Push to talk over Cellular (PoC), Instant Messaging and Presence (IMP), shared whiteboards, multiparty conferencing, interactive gaming, and many others. IMS is capable of performing advanced session control over these applications on different ways:

- *Customizing the application depending on the user equipment or the access network.* For example the network could reduce the signal bitrate wherever the access network provides limited bandwidth to the subscriber or resize the image in a video telephony application when the video is displayed in the small screen of a mobile phone.
- *Modifying sessions in accordance with user preferences.* For example, some subscribers may want to configure their mobile phones to switch to Wi-Fi mode in the office and 3G mode in the car or the street. Other users may want to display the video signal in a video conference only when they have wireline access in order to optimize bandwidth usage.
- *Allowing roaming users to access to services located in the home network from the visited network.* An example of this situation is a user accessing to an IPTV channel provided in this home network from a remote network.

The outstanding features of IMS makes it a key requirement for FMC and effective IP-centric, quadruple play services rollout.

9.3.1 Main Architectural Entities and Interfaces

The IMS is made up by a collection of core network elements implementing sets of standard functions. These network elements are connected between them in normalized reference points (see Figure 9.24).

SIP with 3GPP specific extensions is the core signaling protocol in IMS and therefore many important IMS network nodes are also SIP architectural entities like user agents, proxies or registrars.

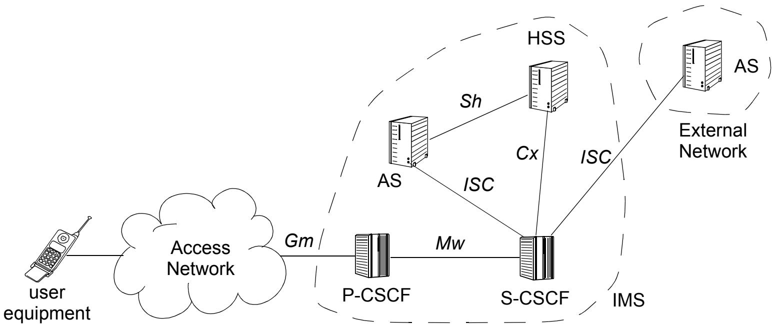


Figure 9.24 Simplified view of the IMS architecture

One of the most important functions within the IMS is the CSCF. This function is implemented by SIP servers. The CSCF interconnects subscribers with services and perform session handling for them. The most important CSCF types are the following:

- The *Proxy CSCF (P-CSCF)* is the first contact point of the user equipment within the IMS. The P-CSCF is a SIP proxy that forwards SIP messages between the user equipment and the right nodes in the IMS. The P-CSCF is also involved in authorization of bearer resources, QoS management and securing communications between the user equipment and itself. To perform these tasks the P-CSCF needs to be connected to the user equipment (G_m reference point). This connection must support the SIP protocol.
- The *Serving CSCF (S-CSCF)* is the function that provides services to subscribers. The S-CSCF behaves like a SIP registrar. It accepts or rejects registration requests from the user equipment. Once authenticated, subscribers are authorized to access network resources depending on their particular profiles. The S-CSCF may also behave as an ordinary SIP proxy forwarding SIP requests or replies towards the right destination endpoint. The S-CSCF is connected to the P-CSCF in the M_w reference point. This point supports the SIP protocol.

A P-CSCF and a S-CSCF is designated for every subscriber during registration in the IMS enabled network. Splitting the CSCF among different network nodes enables support of roaming users (see Figure 9.25). Services are always provided from the subscriber's home network. If the subscriber is in a roaming scenario, the visited network selects a P-CSCF for that user that acts as the contact point of the user with the IMS. If the user is in a non-roaming scenario, then the P-CSCF is still necessary but in this case it can be selected by the own home network.

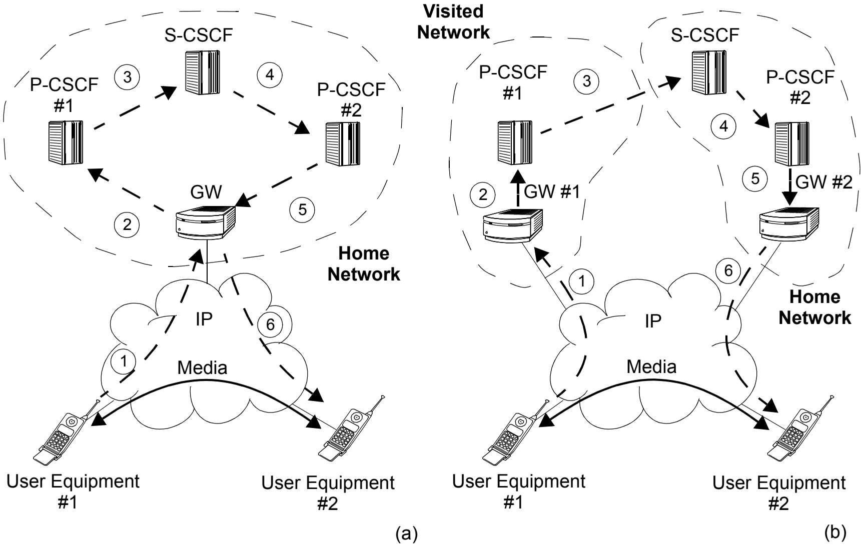


Figure 9.25 Simplified diagram of the network elements and information flows implied in session establishment for roaming and non-roaming users in IMS. (a) Conference in a non-roaming scenario. The media goes from origin to destination through the gateway GW. The SIP signaling traverses the P-CSCFs designated for source and destination subscribers. The common S-CSCF routes signaling between P-CSCFs. (b) Conference in a roaming scenario. The media goes from origin to destination through the gateways GW #1 and GW #2. The signaling traverses the P-CSCF #1 designated by the visited network for user equipment #1 and the P-CSCF #2 designated by the home network for user equipment #2. The common S-CSCF routes signaling between P-CSCFs

The IMS can provide custom services to every subscriber depending on his or her profile. To do that, the IMS must store users' identities along with their subscription information, including services that they are allowed to use and servers and networks they are allowed to access. All these data are held in the HSS. The HSS is also useful to keep centralized information about which particular user is connected to the network, who is registered at a given moment, and which is his or her physical location. In order to allow the S-CSCF to authenticate users, authorize usage of network resources, and route signaling messages to the right destination, it needs to access to the HSS. This connection is done at the C_x reference point. The C_x interface supports the Diameter protocol, specifically designed for Authentication, Authorization and Accounting (AAA) tasks. The HLR database is closely related with the HSS. In fact, the HLR provides a subset of the functions of the HSS.

Large networks may contain several HSSs. In such cases it may be necessary to implement a Subscription Locator Function (SLF) in the IMS. The SLF is in charge of locating the HSS containing the data wanted by other network elements like a S-CSCF. The S-CSCF and the SLF are connected at the D_x reference point that supports the Diameter protocol.

The IMS hosts services in Application Servers (ASs). Examples of services hosted by ASs are presence, messaging, shared whiteboard and many others (see Paragraph 9.3.2)

The IMS is designed to tightly control the operation of networks that carry user information but IMS does not carry user data itself. IMS gateways are divided in a control entity and a enforcement entity. Bearers traverse the enforcement entities but they do not need to go through the management elements of the IMS (see Figure 9.26). Control and enforcement entities have the following role:

- The *control entity* interpret information coming from an AS and S-CSCF and controls the enforcement entity accordingly.
- The *enforcement entity* follows the control commands coming from the management entity and provides to it resources to be controlled.

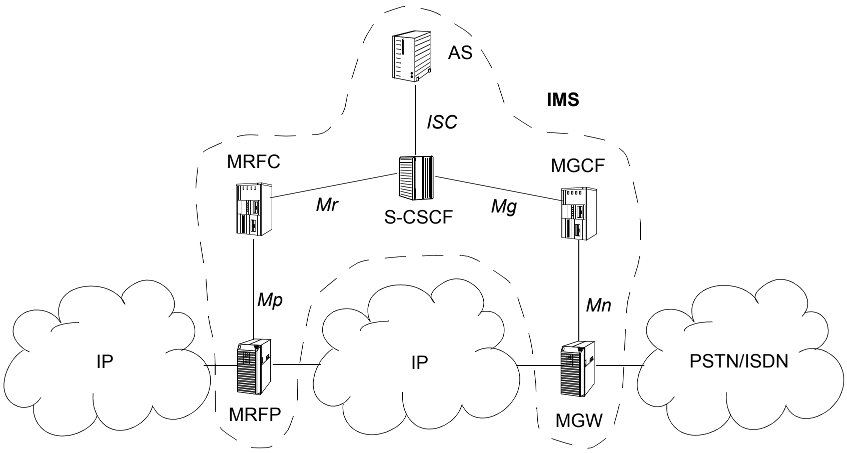


Figure 9.26 The IMS controls user data flows but it does not transport user data itself. Gateways are divided in management entities (MRFC, MGCF) and enforcement entities (MRFC, MGCF)

The Multimedia Resource Function Processor (MRFP), is an enforcement entity that performs operations on media streams in an IP network such as stream mixing and audio or video transcoding. The MRFP is controlled by the Multimedia Resource Function Controller (MRFC) following instructions from ASs and S-CSCFs. The interface between the management and the enforcement entities (M_p reference point) is compliant with Megaco/H.248.

Interworking with PSTN/ISDN is based in the same concept. In this case, the enforcement entity is a Media GateWay (MGW). The MGW terminates bearer channels in the PSTN/ISDN network and media streams in the IP network. It may need to perform transcoding functions on user data as well. Call control is performed by a Media Gateway Controller Function (MGCF). The MGFC controls the media translations performed by the MGW by means the M_n interface based on Megaco/H.248. The S-CSCF must interact with a MGCF when a user desires to communicate with an entity connected to the PSTN/ISDN to do that, SIP is used. The MGCF translates SIP messages to SS7 signaling messages used by the circuit switched network. To do this task, the S-CSCF is assisted by the Breakout Gateway Control Function (BGCF). The BGCF finds the network in which breakout is to occur and chooses the right MGFC there.

9.3.2 Services

Services in the IMS are hosted by entities known as Application Servers (ASs). ASs may be provisioned by the network operator that owns the IMS-enabled network or by an external service provider. In fact, ASs are envisioned as the way service and content providers will interact with subscribers in the IMS environment. With the help of this new approach, service providers and subscribers will have at their disposal IMS advantages like guaranteed QoS. Trust relationships between IMS network elements owned by the network operator and ASs owned by service providers greatly simplifies charging to subscribers for the services. Users do not need to log every time they need to access a new service, only a single registration procedure with the IMS is necessary. These features, known as unified billing and single sign on are two key advantages of IMS for service providers.

On the other hand, the IMS operator gets closer control of network resources and shares revenues with service providers. IMS and non-IMS provision models can coexist. However, IMS service providers have competitive advantage against non-IMS service providers.

Within the IMS, the S-CSCF is connected to the ASs by means the IMS Service Control (ISC) interface that is SIP based. ASs not based on SIP can also be connected to the IMS with the help of dedicated gateways (see Figure 9.27). Interworking with Customized Applications for Mobile networks Enhanced Logic (CAMEL), and Open Services Architecture (OSA) ASs are currently supported by the 3GPP standards.

- *Customized Applications for Mobile networks Enhanced Logic (CAMEL)* allow operators to define value added services equivalent to the Intelligent Network (IN) services available for fixed networks but for GSM and UMTS mobile networks. CAMEL is designed to enable service provision even for roaming users. CAMEL standards were released in 1997 for first time and have been improved several times so far. CAMEL services are accessible to IMS subscribers through a gateway implementing the Service Switching Function.
- *The Open Services Architecture (OSA)* is a joint initiative of the 3GPP, ETSI, Telecoms & Internet converged Services & Protocols for Advanced Network (TISPAN), and a multi-vendor consortium known as Parlay Group. The OSA standard defines a framework that makes the functionality of 3G mobile networks accessible to service application developers in a standard, flexible and scalable way. OSA is an advantage to service developers, since it abstracts them from the complexity of the underlying network infrastructure and low lever telecommunication protocols, allowing them to focus on the creation of the service logic. OSA applications are accessible to IMS users through an OSA Service Capability Server.

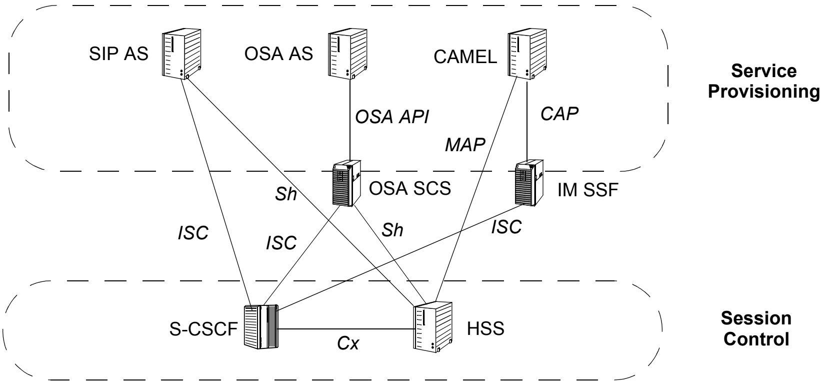


Figure 9.27 In IMS, services are provided by means ASs connected to the S-CSCF. SIP ASs exchange information directly with the S-CSCF in the ISC interface. Other ASs rely on special gateways that translate information into ISC standards. AS can also directly access to the information stored in the HSS by means the S_h interface.

Currently, definition of interoperable services for IMS depends on an organization that groups equipment vendors, mobile operators and software companies known as Open Mobile Alliance (OMA). The OMA was created on June 2002 and its activity is essential for global adoption of mobile services. Specifications maintained by the OMA define requirements for Multimedia Messaging Service (MMS), Digital Rights Management (DRM), Instant Messaging and Presence (IMP), Push to talk over Cellular (PoC), gaming services, broadcast services, and many others.

An important example of IMS service is Voice Call Continuity (VCC), introduced in the Release 7 of the 3GPP specifications. VCC enables transparent handoff of the voice service for any access network. VCC works for mobile or wireline access networks and for circuit switched or packet switched modes. However handoff of multimedia streams are out of the scope of the current specification.

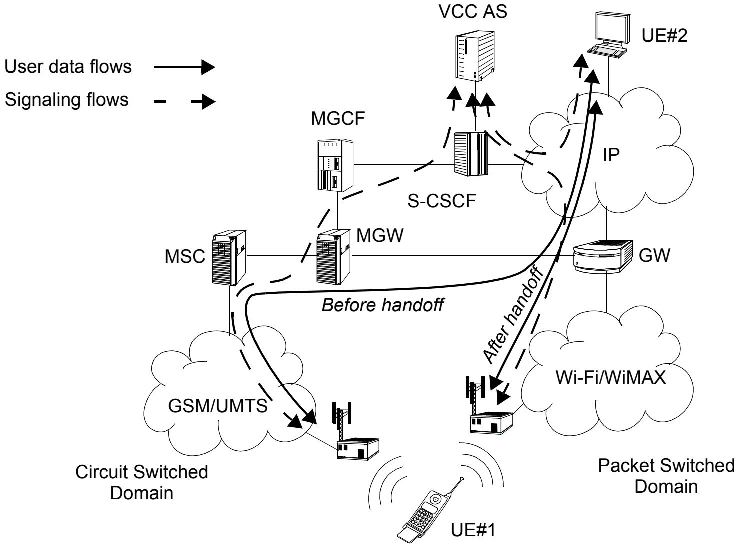


Figure 9.28 Simplified operation of VCC. The UE#1 voice call is transferred from the UMTS or GSM radio access network to a Wi-Fi or WiMAX access network. In the example, the receiver, identified as UE#2 is operating in packet switched mode and is therefore receiving VoIP data.

The VCC service is provided by an special IMS AS. Under request of a dual user equipment, the VCC AS generates signaling to redirect voice traffic from the origin to the destination access networks (see Figure 9.28). Like other IMS ASs, the VCC AS is a SIP server. That means that it requires help of signaling gateways to interact with non-SIP devices. To operate with the GSM/UMTS circuit switched domain, the IMS uses the MGCF to translate SIP messages to the SS7 commands that can be interpreted by the MSC.

9.3.3 User Identity

In IMS as well that in GSM and UMTS networks, user identification is independent of identification of user terminals. When compared with the PSTN, the concept of user identification is new because the PSTN identifies telephones rather than users.

User identification adds personal mobility to terminal mobility. Users can theoretically attach their smart cards to any mobile phone. The network stores the user identities in the HLR or HSS along with their profiles and many service preferences. During registration, the subscriber information from the smart card is authenticated rather than the user equipment itself. This allow subscribers to log in the network from different devices and enjoy a static environment (including the telephone number).

Subscriber identities IMS networks are very different to GSM and UMTS identities. Networks that are IMS and GSM/UMTS at the same time should provide to their subscribers GSM/UMTS identities as well as IMS identities.

9.3.3.1 User identity in GSM and UMTS

Users in GSM and UMTS networks are identified by their International Mobile Subscriber Identity (IMSI), a unique, non-dialable number of up to 15 digits. The IMSI is made up by three parts: A three-digit Mobile County Code (MCC), the Mobile Network Code (MNC) that consists on two digits (three digits in the US) and the remaining digits are the Mobile Subscriber Identity Number (MSIN). The IMSI is stored in the HLR and in a smart card handed out to the subscriber. An example of a valid IMSI is 262013564857956. Here the MCC is 262 (Germany), the MNC is 01 (T-Mobile D) and the MSIN is 3564857956.

In addition to the IMSI, there are other numbers associated with subscriber and user equipment identity:

- The *Temporary Mobile Subscriber Identity (TMSI)* replaces the IMSI in most of the transmissions between the user equipment and the network. It is a 32 bit number assigned by the network when the mobile is switched on. Moreover, the network may change the TMSI of the mobile under various circumstances to avoid the subscriber from being identified and tracked by eavesdroppers on the radio interface.
- The *International Mobile Equipment Identity (IMEI)* is a unique 15 digit number that identifies a GSM or UMTS user equipment. The IMEI identifies only the user equipment and it has no relationship with the subscriber identity. This identifier can be used by a network operator to offer custom services to subscribers depending on their user equipment. The IMEI has been also used to prevent stolen user equipment from being used. Stolen mobile phones can be blacklisted and theoretically left unusable. An example of a valid IMEI number is 356057-00-481557-7.
- The *Mobile Station Integrated Services Digital Network (MSISDN)* is the name given by the 3GPP to the telephone number associated to a subscriber. The MSISDN conforms with ITU-T E.164 and should comply to each country's

ISDN numbering plan. MSISDN numbers consists in a Country Code (CC), a National Destination Code (NDC) and a Subscriber Number (SN). An example of a valid MSISDN is 34659057404.

9.3.3.2 User identity in IMS networks

IMS subscribers are identified by means one or various private user identities and one or various public user identities:

- A *private user identity* can be thought as a user subscription identifier. In other words, there are as many private user identities as subscriptions to the IMS network the user has. The private user identifier uses a username@realm syntax as defined in RFC 2486. Storage of the private user identity in a subscriber's smart card and in the HSS is necessary to user's subscription authentication during the registration process. The private user identity is used for transactions that may have security concerns (for example, registration, re-registration and de-registration) but it is never used for routing of SIP messages. That means, that one user does not need to know other users' private identities to communicate with them.
- A *public user identity* identifies the user rather than the user subscription. One user may have several public identities for different purposes (see Figure 9.29). It is also possible to share one or more public user identities among various private user identities. SIP URIs like "sip:alice@albedo.biz" or tel URIs like "tel:+44-1628-503500" are both accepted for public user identity specification. Public user identities does not need to be authenticated during the registration process and their main purpose is to allow other users or services to reach the subscriber.

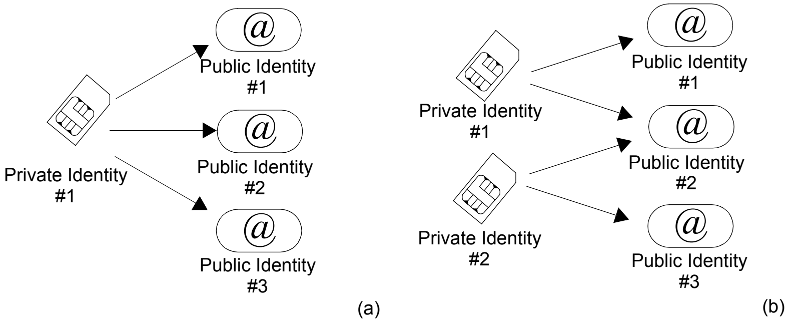


Figure 9.29 Relationship between private and public user identities in the IMS. (a) a single private user identity is associated with three public user identities. (b) two private user identities share the public user identity #2

Public and private user identities are not changeable by the user. When a public user identity is shared among various user equipment the subscriber can be reached at the same time in different devices like for example a PDA and a computer. Some user equipment is able to select which user equipment they want to communicate with, by means Globally Routable User agent URIs (GRUUs). A GRUU is an entity that is made up by a unique combination of a public user identity and a code that identifies the user agent. A GRUU can be public or temporary. Public GRUUs expose the identity of the user and temporary GRUUs hide the user identity for privacy purposes. GRUUs are SIP URIs with the special *gr* parameter. A public GRUU may look like "sip:alice@albedo.biz;gr=kjh29x97us97d" while a temporary GRUU looks like "sip:8ffkas08af7fasklzi9@albedo.com;gr".

9.3.4 AAA with Diameter

Authentication, Authorization, and Accounting (AAA) are three related terms that can be defined as follows:

- *Authentication* is the act of verifying the identity of an entity like for example the subscriber of a service provided by a network or a remote server that requests an specific network resource.
- *Authorization* is the act of determining whether a service subscriber or any other requesting entity will be allowed to access to a resource provided a network or a server.
- *Accounting* is the act of collecting information of a resource usage for the purpose of capacity planning, auditing, billing or cost allocation.

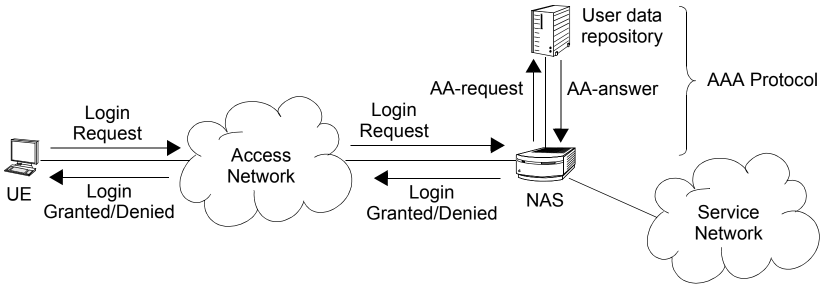


Figure 9.30 NAS application for providing access to subscribers to network resources. The NAS is an IP router that authenticates users and authorizes access to the network resources. However, user subscription data is stored in the user data repository and the NAS needs to check this information. Communication between the NAS and the user data repository is done with an AAA protocol like RADIUS or Diameter

To perform AAA, the system need to know the resources that every user is allowed to access. This information is part of the user subscription data and in small networks it can be stored in the own server that provides access to the network. However, this solution becomes unpractical for medium and large networks. When this happens, it is usually better to store user subscription information in a dedicated database and allow the access servers to read this information from the database. This solution provides enhanced security, allows users to log from different access servers without replicating user credentials, and keeps track of user information records from a central location. The protocol that communicates the access servers and the user subscription database is an AAA protocol like the Remote Authentication Dial-In User Service (RADIUS) protocol (see Figure 9.30).

Within the IMS, AAA information exchange is performed by the Diameter protocol. Specifically, Diameter is the basis of the C_x interface that allows information exchange between the HSS and different types of CSCFs. Diameter, as specified in RFC 3588, is a replacement for the classical RADIUS. RADIUS was initially used to provide PPP dial-up access to network resources by means a Network Access Server (NAS). However, it has been necessary to develop a new AAA protocol to take into account the requirements of new access technologies like wireless cellular networks.

Table 9.5
Comparison between the Diameter and RADIUS AAA protocols

	RADIUS	*Diameter*
Transport Protocol	UDP	TCP and SCTP
Security	Hop-to-Hop	Hop-to-Hop, End-to-End
Peer discovery	Not supported	Supported
Server initiated transactions	Not supported	Supported
Maximum attribute size	255 bytes	16,777,215 bytes
Vendor-specific Support	Vendor specific attributes	Vendor specific messages and attributes
Capabilities negotiation	Not supported	Supported

Diameter has been derived from RADIUS but the former provides many advantages when compared with the latter. RADIUS operates over the unreliable UDP transport protocol but Diameter works either with TCP or the Stream Control Transmission Protocol (SCTP) on port 3868. Both TCP and SCTP provide reliable communications and they are more suited to exchange sensitive data transported in AAA applications than UDP. Moreover, the SCTP has been specifically designed to transport signaling messages over IP networks and provides even more benefits to AAA applications than TCP, like for example path selection and monitoring or protection against Denial of Service (DoS) attacks. In fact, use of SCTP in IMS is mandatory. Other advantages of Diameter over RADIUS are a new End-to-End security framework, automatic peer discovery capability, better support of vendor specific messages, improved negotiation of capabilities during session startup, support of longer attributes in messages and possibility to start transactions from the Diameter server and not only from the client (see Table 9.5).

Diameter has been conceived as a peer-to-peer protocol. That means that with Diameter, the communicating parties may act both as clients or servers and they may change their role in different transactions. In addition to Diameter clients and servers, there are Diameter agents. There are different types of Diameter agents (see Figure 9.31):

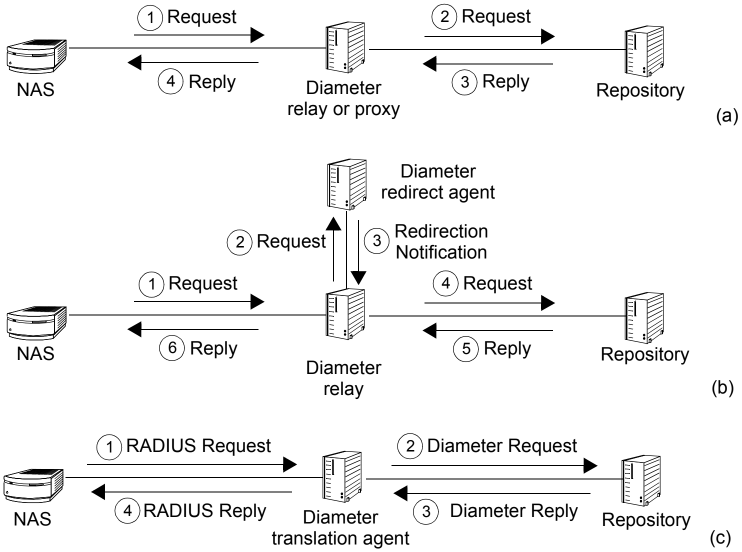


Figure 9.31 Diameter agents: (a) Relays and proxies route messages to a remote location. (b) Redirect agents notify how to reach a destination. (c) Translation agents translate between different AAA protocols.

- *Relay Agents and Proxies* forward Diameter messages to the appropriate destination. This is useful in large networks to aggregate messages from different access servers to a central node. The difference between relay and proxy agents is that proxies are allowed to modify messages with different purposes but relay agents are only allowed to append or remove service layer routing information to Diameter messages but they do not modify other portion of them.
- *Redirect Agents* do not relay Diameter messages, they rather notify to other agents how to reach a destination.

- *Translation Agents* translate messages between one AAA protocol to another. For example a translation agent may translate RADIUS messages into Diameter messages.

Diameter messages carry sensible information. For example an operator may charge its subscribers by means an application based on Diameter. In order to provide proper confidentiality, authentication and integrity, Diameter relies in features provided by the own Diameter and other external protocols. End-to-end security is optional and provided by an extension of the base Diameter protocol. Hop-by-hop protection is based either in IP security (IPsec) or Transport Layer Security (TLS). However, for IMS the use of IPsec in the network layer is mandatory. Specifically, IMS standards mandate that the IPSec Encryption Security Envelope (ESP), as defined in RFC 2406, must be used for all inter-domain signaling traffic to achieve confidentiality, integrity, and data source authentication. Peer authentication at session startup, negotiation of security associations and secure key management in IMS use the standard mechanisms defined by the Internet Key Exchange (IKE) as per RFC 2409.

Table 9.6
Authentication and Authorization IMS commands defined in TS 29.229

Command Name	*Issued by*	*Description*
User-Authorization-Request (UAR)	Client	Requests authorization for registration of a user.
User-Authorization-Answer (UAA)	Server	Response to a UAR command.
Server-Assignment-Request (SAR)	Client	Requests storage of the name of the server that is currently serving a user.
Server-Assignment-Answer (SAA)	Server	Response to a SAR command.
Location-Info-Request (LIR)	Client	Requests the name of the server that is currently serving a user.
Location-Info-Answer (LIA)	Server	Response to a LIR command
Multimedia-Auth-Request (MAR)	Client	Requests security information
Multimedia-Auth-Answer (MAA)	Server	Response to a MAR command
Registration-Termination-Request (RTR)	Server	Requests de-registration of a user
Registration-Termination-Answer (RTA)	Client	Response to a RTA command
Push-Profile-Request (PPR)	Server	Updates the subscription data of a user
Push-Profile-Answer (PPA|)	Client	Response to a PPR command

Peer authentication and authorization in Diameter sessions must not be confused with subscriber authorization and authentication (see Paragraph 9.3.6.1). In fact, the necessary commands for subscriber authentication and authorization are not included in the base Diameter specification. Authentication and authorization may change among applications and thus these functions are defined in extensions of the basis protocol not included in the main standard. For example, RFC 4005 defines a

Diameter application for authenticate and authorize users in NAS scenarios and RFC 4740 is a Diameter application for user authentication and authorization performed by SIP network entities. The 3GPP defines its own authentication and authorization application for IMS in TS 29.229 (see Table 9.6).

9.3.5 Policy and Charging Control

Charging and QoS control are different but related topics: In a network, users pay to use transmission resources that can be specified by a set of QoS metrics. Policy control and charging control were separate topics in old 3GPP releases. However, they have been unified in 3GPP Release 7. The current Policy and Charging Control (PCC) framework is a joint evolution of the Session-Based Local Policy (SBLP) and the Flow Based Control (FBC) framework.

- As defined in Release 5 of 3GPP specifications, SBLP allows the IMS to control and manage the IP network resources devoted to an specific application. Specifically, the SBLP framework allows to coordinate the QoS advertised during session setup (for example using an SDP body in an SIP INVITE transaction) and the actual QoS reserved in the bearer domain. Without this kind of coordination and control, user equipment would be able to reserve and use more resources than advertised during session setup without being charged for it. The Policy Decision Function (PDF) is a key entity of the SBLP framework. The PDF authorizes specific service levels for different applications and users taking as a basis subscriber profiles and information SIP/SDP control plane information.
- The FBC is defined in the Release 6 of 3GPP specifications. The FBC framework solves the problem of charging at the bearer layer with flow granularity rather than with session granularity. Thanks to the FBC it becomes feasible charging individual SIP VoIP calls using SIP signaling generated by the communicating parties, charge individual FTP data flows and web services between specific subscribers and servers, and many other alternatives. Furthermore, FBC enables the IMS operator to use different charging models like volume based charging, time based charging, event based charging, and combinations of them. An important element in the FBC framework is the Charging Rules Function (CRF). This entity generates charging rules for the appropriate network elements where charging occurs.

In the new PCC framework, there is a single entity that generates charging rules and allocates network resources. This unified entity, known as Policy control and Charging Rules Function (PCRF), generates PCC rules that are used at the same time for policy and charging purposes. PCC rules are applied to an enforcement point known as Policy and Charging Enforcement Function (PCEF), that is usually integrated in a media gateway such as a GGSN (see Figure 9.32). The interface between the PCEF and the PCRF is based on Diameter as specified in RFC 3588 with appropriate extensions. The PCEF has the ability to process and filter the user plane traffic to ensure that policy defined by the PCC rules is correctly applied. To do that, the PCEF may discard or delay user traffic or may also deny resource reservations carried out by user equipment. The PCEF also generates charging information when a so called chargeable event occurs, that is, user activities that utilize or consume network resources. Charging information is then processed either by the Offline Charging System (OfCS) or the Online Charging system, depending on the agreement with the subscriber. Offline and online charging are defined as follows:

- *Offline charging* is a billing mechanism where charging information does not affect, in real time the service. Charging information is delivered to the operator after the resource usage occurred and as a result interaction with the service is not possible.
- *Online charging* involves real-time interaction between the charging framework and the service to accomplish credit-control or budget supervision. Authorization for resource usage must be obtained by the network prior to the actual resource usage to occur. Resource usage permissions may be limited in its scope. For example permission could be granted for a limited amount of data or time.

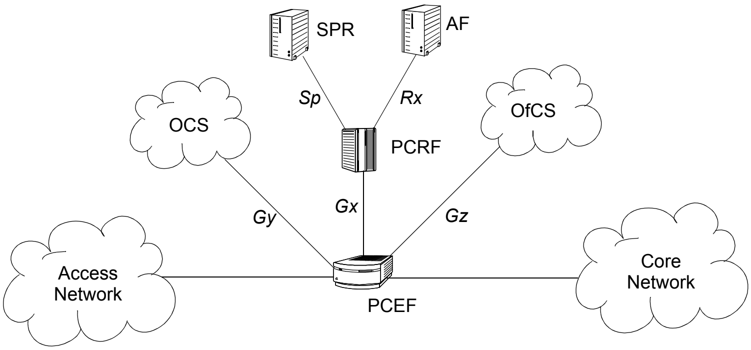


Figure 9.32 PCC architecture for IMS.

The OfCS and OCS generate charging records that are delivered to a Billing Domain (BD), an operator's postprocessing system to process and store charging information. The PCEF and the OCS are connected through the G_y reference point that is Diameter based and uses the Diameter Credit-Control Application defined by RFC 4006. Connection between the PCEF and the OfCS is done by means the G_z reference point that is Diameter based as well and it is defined in 3GPP TS 32.240.

To generate PCC rules, the PCRF receives dynamic session information information from an Application Function (AF) that is usually represented by the P-CSCF designated for the subscriber. Examples of information that can be used for charging and policy control are registration in the IMS, a SIP call origination or release, or many others. Generation of PCC rules may also depend on the user profiles. For this reason the PCRF is allowed to retrieve information from the database where profiles are stored. Within the PCC framework, this database is known as Subscription Profile Repository (SPR). The PCRF is communicated with the AF by means R_x reference point and with the SPR with the SPR by means the S_p reference point. Both are based on the Diameter protocol.

9.3.6 Basic Procedures

IMS procedures include registration and de-registration, call setup and termination, messaging and many others. These procedures are based in dialogs between IMS network elements. Most of them, use SIP and Diameter.

9.3.6.1 Registration

Before being served, users must register in the IMS network. However, there are at least three basic conditions that must be fulfilled before registration.

1. Establishment of a bearer channel in the access network is necessary to enable information exchange with core nodes. Sign on within these access networks may be necessary before the IMS registration. For example, GSM and UMTS technologies need to authenticate users before allowing them to use network resources.
2. IMS registration is a SIP-based procedure that needs IP connectivity to work. Therefore, the user equipment must receive at least a valid IP address that will allow them to communicate with remote IP peers. Some networks have special procedures to assign IP addresses to users. For example, GPRS networks can do that in the PDP context activation procedure. The DHCP protocol constitutes a generic way to assign IP addresses to users in networks without specific procedures to do that.
3. The user equipment needs the IP address of at least one P-CSCF to establish communication with the IMS core. The address can be retrieved during the IP connectivity establishment phase if the access network provides such means. Otherwise, a generic procedure based on the DNS and DHCP protocols can be used. In the generic procedure, the user equipment requests to a DHCP server the IP address of one or several DNS servers and the domain name of at least one P-CSCF. The user equipment can then resolve the domain names of the P-CSCFs by means one or various DNS queries. The P-CSCF does not need to be in the home network. P-CSCFs placed in visited networks are used in

roaming scenarios.

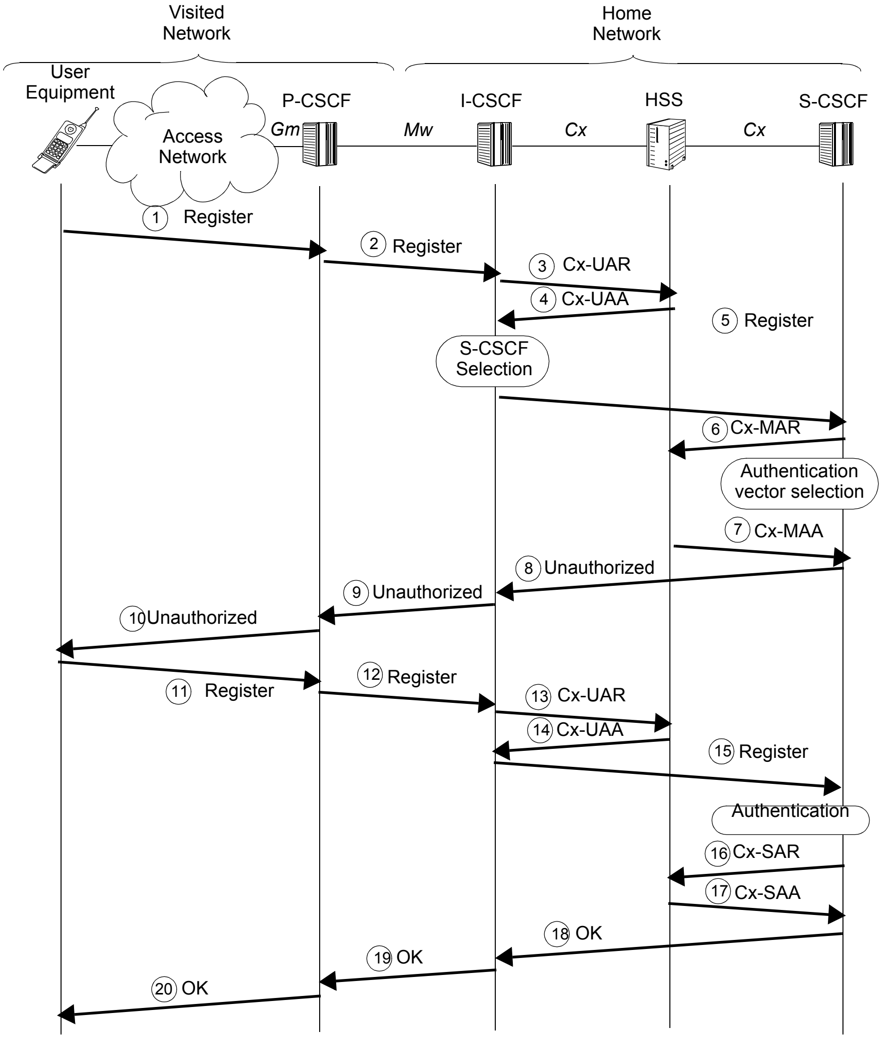


Figure 9.33 Typical registration information flow in an IMS network. The user equipment is considered to access to the home network through an external visited network.

Once the user equipment has the IP address of a P-CSCF, registration in the IMS network can start. Registration involves mutual authentication of the subscriber and the network. The mechanism defined for this purpose is known as IMS Authentication and Key Agreement (IMS AKA). IMS AKA is a challenge-based procedure derived from the UMTS authentication (UMTS AKA). IMS AKA dialogs are implemented with SIP and SIP 3GPP specific extensions. Furthermore, extra security is needed to protect signaling between the user equipment and the P-CSCF. Setup of Security Associations (SA) between these network elements also uses SIP and follows RFC 3329.

The following sequence illustrates the registration sequence for a roaming user (see Figure 9.33):

1. The user equipment sends a SIP REGISTER request to the P-CSCF. This message contains public and private IMS user identities, home network domain name and other information.
2. IMS services are always provided from the home network. For this reason, the P-CSCF must forward the register information to the user's home network. To do that, the P-CSCF examines the home network domain name included in the register request. The registration information flow reaches an special CSCF in the home network know as Interrogating CSCF (I-CSCF). This server acts as the entry point of signaling flows from external IMS networks. The IP address of the I-CSCF is published in DNS servers to enable roaming users to reach this server using the home server domain name.
3. The I-CSCF issues a Diameter query in the C_x interface to the HSS to request whether the user is allowed to register in the network through this P-CSCF or not. The Diameter message User Authorization Request (UAR) defined in 3GPP TS 29.229 can be used for this purpose.
4. The HSS determines whether the user is allowed to register and answers the query from the I-CSCF. The answer to the UAR query is the User Authorization Answer (UAA). Sometimes the answer from the HSS contains the domain name of the S-CSCF designated for the user, sometimes it only contains information about the S-CSCF that enables the I-CSCF to perform the right choice.
5. The I-CSCF determines the IP address of the S-CSCF designated for the user and resends the register information flow to that server.
6. The S-CSCF challenges the user before allowing registration. To do that, the S-CSCF may need to request IMS AKA authentication vectors from the user profile stored within the HSS. The Diameter Multimedia Authorization Request (MAR) message is used for this purpose.
7. The HSS retrieve authorization information for the user to be authenticated and delivers it to the S-CSCF by using a Diameter Multimedia Authorization Answer (MAA) message.
8. The S-CSCF uses the authentication information received from the HSS to compute a challenge for the user. Then it attaches the challenge within a 401 Unauthorized SIP response. The response is delivered to the I-CSCF.
9. The 401 Unauthorized response propagates through the signaling path and it arrives to the P-CSCF

10. The P-CSCF receives the 401 Unauthorized SIP message and resends it to the user equipment.
11. Upon receiving the 401 Unauthorized response, the user equipment generates a new SIP REGISTER request. This time it calculates the challenge response and attaches it in the new REGISTER message before being sent.
12. The P-CSCF receives the new REGISTER request and forwards it to the user's home network via the I-CSCF.
13. The I-CSCF queries again the HSS for information about the subscriber registration status. The Diameter UAR message is used for this purpose.
14. The HSS replies to the I-CSCF with the requested information by means the Diameter UAA message.
15. The I-CSCF forwards the REGISTER message with the challenge response to the S-CSCF.
16. The S-CSCF receives the REGISTER request from the user and it authenticates it. If the result of this operation is successful, it issues a query to the HSS so that the HSS identifies it as the service control server for that user. The Diameter Server Assignment Request (SAR) is used for this purpose.
17. The HSS bounds the user identity to the S-CSCF included in the Diameter query and issues a response. The response message to a SAR is Server Assignment Answer (SAA). The answer may include names of platforms which can be used for service control while the user is registered and other information.
18. If the S-CSCF is able to accept the registration request it delivers a SIP OK response to the I-CSCF.
19. The I-CSCF resends the information flow to the P-CSCF in the visited network.
20. The information flow reaches the user equipment. Starting from this moment the user becomes registered in the IMS network.

9.3.6.2 Call Origination

Like other IMS procedures, call origination is based in SIP with extensions specifically designed by 3GPP for call control (see Figure 9.34). A user who wants to establish a call with a remote user issues a SIP INVITE to the recipient. The INVITE request contains an SDP body with descriptions of the media formats and codecs to be used during the session. The SDP body may also contain descriptions of the required QoS for the call. The INVITE request with the initial SDP description is propagated through the IMS network. Messages from roaming users are routed through

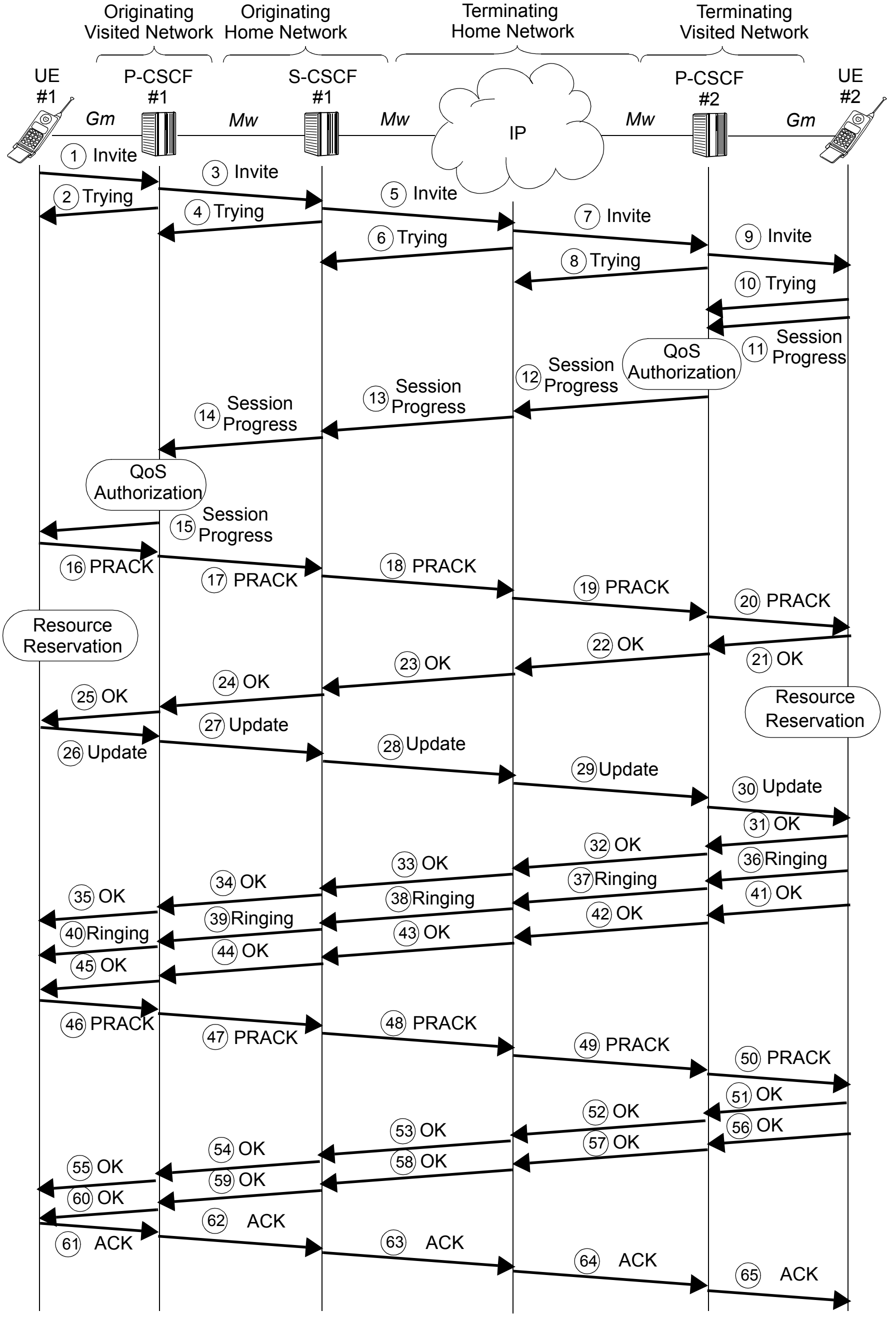


Figure 9.34 Typical session establishment signaling flow in IMS (roaming case).

an I-CSCF to the S-CSCF that is always in the home network. As the recipient is potentially a mobile user, the S-CSCF will need to request location information for that user from the HSS. The Diameter message Location Info Request (LIR) is used for this purpose.

When the called user equipment receives the initial SIP INVITE it starts a session negotiation stage using provisional SIP requests and responses and SDP. Specifically, the called user equipment responds with a 183 session progress to the initial INVITE and the originator user equipment uses the PRovisional ACKnowledge (PRACK) procedure to send provisional information. During this stage two important things happen:

- Both communication ends agree the media flows to be used for the session and the media codecs to be used for every media flow. Media flow and codec negotiation may require several SDP offers before an agreement is reached.
- During the negotiation stage, the P-CSCFs authorize QoS resources necessary for the call.

Media flow and codec negotiation finishes with a 200 OK reply. After delivery of the last negotiation message, the communication ends start the resource reservation procedure. Actual reservation depends on the particular network infrastructure but is should agree with the QoS resources allocated by the P-CSCF. Otherwise, resource reservation may be denied. The call originator uses an SDP body sent within a SIP UPDATE request to inform to the remote end that QoS resources have been successfully allocated. The call recipient does the same by replying with a 200 OK reply to the UPDATE request. This reply also uses the SDP protocol to communicate successful resource reservation.

When bearer resources are ready for communication, the call recipient may perform alerting by issuing a 180 Ringing provisional response to the previous UPDATE request. This response is propagated through the signaling path like the previous SIP messages. When the call originator receives the 180 Ringing response it issues a new PRACK request that is answered by a new 200 OK reply.

If the call recipient decides to answer the call, a final OK reply is issued and delivered towards the call originator. The originator then opens the media flows for the calls generates an ACK procedure that is propagated through the signaling path toward the call recipient. The media session can then start.

9.3.7 The Next Generation Network

The concept of an integrated broadband network has developed over the last years and has been labeled Next Generation Network (NGN). Requirements, features and implementation of the NGN depend on who defines it. With time, the concept of

NGN has become rather imprecise. The ITU-T defines an NGN in Recommendation Y.2001 as “a packet-based network able to provide telecommunication services and able to make use of multiple broadband, QoS-enabled transport technologies, and in which service-related functions are independent from underlying transport-related technologies. It enables unfettered access for users to networks and to competing service providers and/or services of their choice. It supports generalized mobility which will allow consistent and ubiquitous provision of services to users”

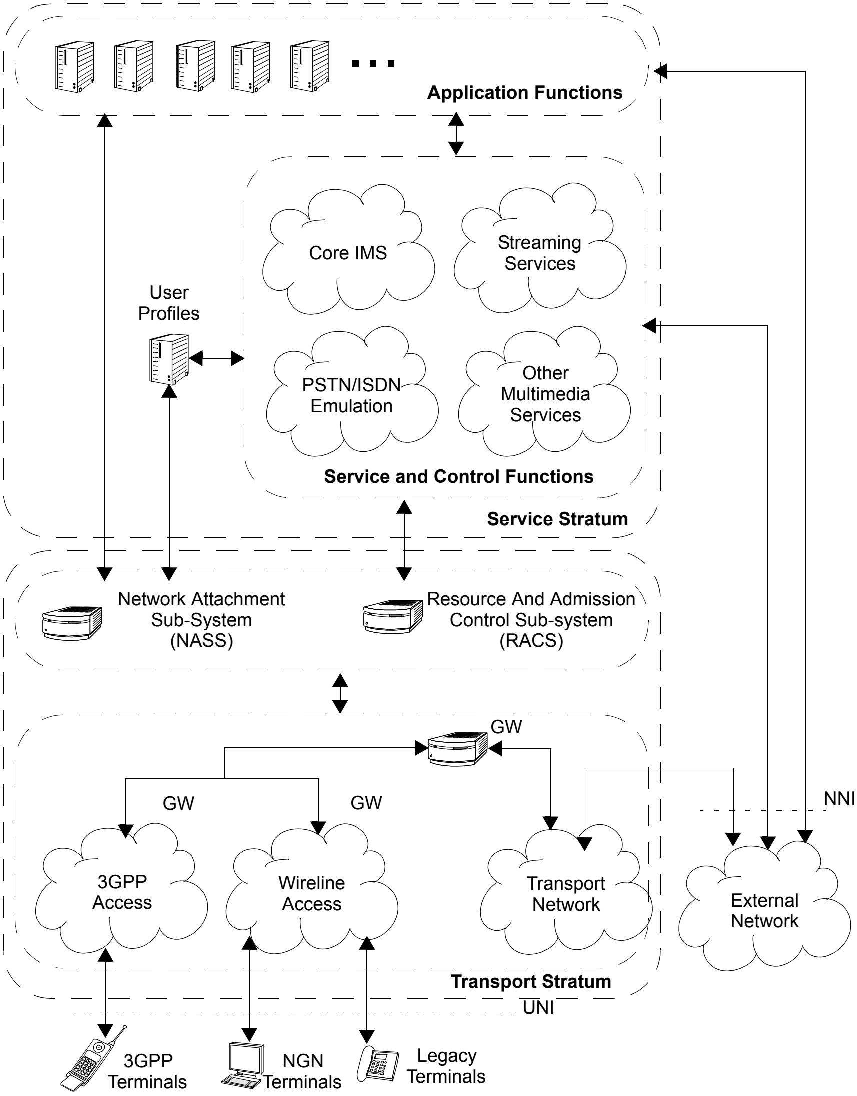


Figure 9.35 TISPAN NGN architecture

Taking this definition as the starting point, specification work of the NGN has started. The organizations more closely involved in the standardization process of the NGN are the ETSI Telecoms & Internet converged Services & Protocols for Advanced Network (TISPAN) and the ITU-T.

The ETSI TISPAN is the result of the merge of two ETSI committees in September 2003: The ETSI Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) and the ETSI Signaling Protocols for Advanced Networks (SPAN). The former, defined requirements for seamless interworking between VoIP and PSTN networks and the latter was devoted to adapt to European networks the ITU-T standards for signaling within the Integrated Services Digital Network (ISDN) scope. The ITU-T started work on NGN almost at the same time that ETSI TISPAN was created. Within the ITU-T, the leading group in NGN work is the Study Group 13 (SG13).

The TISPAN is working closely with the 3GPP to define common core for both wireless and wireline networks. As a result of TISPAN work, Release 1 of NGN TISPAN specifications were published in December 2005. IMS is the fundamental ingredient of the TISPAN project. 3GPP IMS was chosen because it provides many of the features expected from the NGN. Specifically, the IMS is compatible with the concept of a multiservice, ubiquous network thanks of being access-network-agnostic and allowing services and transport decoupling. Currently, the 3GPP extends the IMS to meet the requirements of the TISPAN NGN specifications.

As a result of the ITU-T SG13 work, Recommendations Y.2001 and Y.2011 were released in June 2004. These Recommendations constitute the basis for all other ITU-T studies on NGN. Also, in June 2004, the ITU-T launched the Focus Group on NGN (FGNGN) to coordinate all NGN studies taking as a starting point Recommendations Y.2001 and Y.2011. One of the tasks of the FGNGN was globalization of TISPAN work to a worldwide scope. By the end of 2005, the FGNGN finished its work and all relevant results were transferred to SG13 and other groups within the ITU-T.

The main goal of the Release 1 of TISPAN specifications is to provide an extensive architecture for the NGN. Power of the TISPAN NGN framework is exhibited in two important capabilities included in the new standard:

- It enables delivery of services supported by 3GPP IMS to broadband fixed lines. However, Release 1 is not required to support handoff of communication sessions between access networks. This requirement includes handoffs from or to the wireline access network.
- It enables PSTN/ISDN replacement (in whole of in part). Specifically, Release 1 enables PSTN/ISDN emulation and PSTN/ISDN simulation. The former is defined to be the support of legacy PSTN/ISDN terminal equipment and ser-

vices, the later is the provision of services similar to the PSTN/ISDN to NGN voice or other multimedia terminal equipment. The purpose of PSTN/ISDN emulation and simulation is to enable gradual migration from the legacy infrastructure to the NGN.

Wireline access networks exhibit significant differences with 3GPP mobile access networks. For example, location information is fundamentally different in both network classes, resource reservation mechanisms may be unavailable in wireline access, and IPv4 is still in use in many of them. All these topics are addressed by the NGN specification.

TISPAN has defined IMS extensions and profiles for operation of IMS over wireline access networks like DSL, PON or Gigabit Ethernet. TISPAN has also worked on new subsystems for the NGN with the same purpose. Examples are the Network Attachment Sub-System (NASS) and the Resource and Admission Control Sub-system (RACS) (see Figure 9.35). The former is in charge of IP address provisioning, network level user authentication and access network authorization. The latter provides admission control, resource reservation, policy control and NAT router traversal.

Chapter 10

Carrier-Class Ethernet

Incumbent and competitive operators have started to provide telecommunications services based on Ethernet. This technology is arising as a real alternative to support both traditional data-based applications such as *Virtual Private Networks* (VPN), and new ones such as Triple Play.

Ethernet has several benefits, namely:

- It improves the flexibility and granularity of legacy TDM-based technologies. Many times, the same Ethernet interface can provide a wide range of bit rates without the need of upgrading network equipment.
- Ethernet is cheaper, more simple and more scalable than ATM and *Frame Relay* (FR). Today, Ethernet scales up to 10 Gbit/s, and soon it will arrive to 100 Gbit/s.

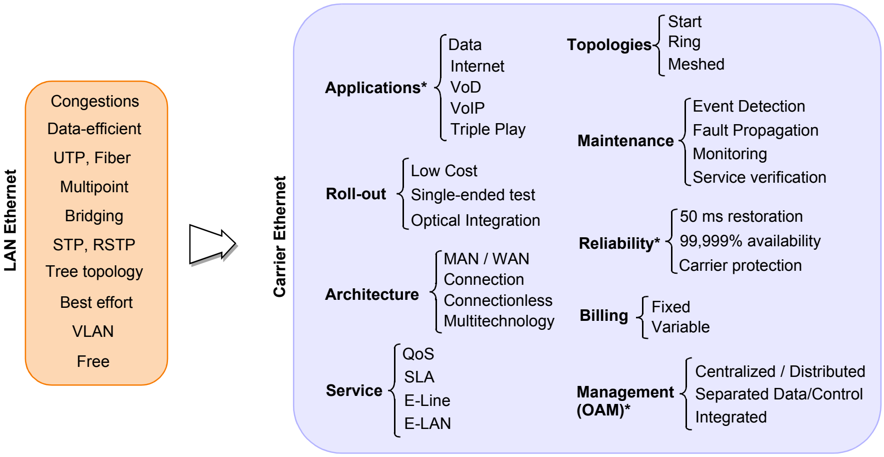


Figure 10.1 The path to Carrier-Class Ethernet

Furthermore, Ethernet is a well-known technology, and it has been dominant in enterprise networks for many years. However, Ethernet, based on the IEEE standards, has some important drawbacks that limit its roll-out, especially when the extension, number of hosts and type of services grow. This is the reason why, in many cases, Ethernet must be upgraded to carrier-class, to match the basic requirements for a proper telecom service in terms of quality, resilience and OAM (see Figure 10.1).

10.1 Ethernet as a MAN / WAN Service

Ethernet has been used by companies for short-range and medium/high-bandwidth connections, typical of LANs. To connect hosts from remote LANs, up to now it has been necessary to provide either FR, ATM or leased lines. This means that the Ethernet data flow must be converted to a different protocol to be sent over the service-provider network and then converted back to Ethernet again. Using Ethernet in MAN and WAN environments would simplify the interface, and there would be no need for total or partial protocol conversions (see Figure 10.2).

Currently, the *Metro Ethernet Forum* (MEF), the *Internet Engineering Task Force* (IETF), the *Institute of Electrical and Electronics Engineers* (IEEE), and the *International Telecommunications Union* (ITU) are working to find solutions to enable the deployment of Carrier-Class Ethernet networks, also known as *Metro-Ethernet Networks* (MEN). This includes the definition of generic services, interfaces, deployment alternatives, interworking with current technologies, and much more. Carrier-Class Ethernet is not only a low-cost solution to interface with the subscriber network and carry its data across long distances, but it is also part of a converged network for any type of information, including voice, video and data.

10.1.1 Network Architecture

The ideal Metro-Ethernet Network makes use of pure Ethernet technology: Ethernet switches, interfaces and links. But in reality, Ethernet is often used together with other technologies currently available in the metropolitan network environment. Most of these technologies can inter-network with Ethernet, thus extending the range of the network. Next-Generation SDH (NG SDH) nodes can transport Ethernet frames transparently (see Paragraph 10.2.3). Additionally, Ethernet can be transported by layer-2 networks, such as FR or ATM.

Today, many service providers are offering Ethernet to their customers simply as a service interface. The technology used to deliver the data is not an issue. In metropolitan networks this technology can be Ethernet or SDH. Inter-city services are almost exclusively transported across SDH.

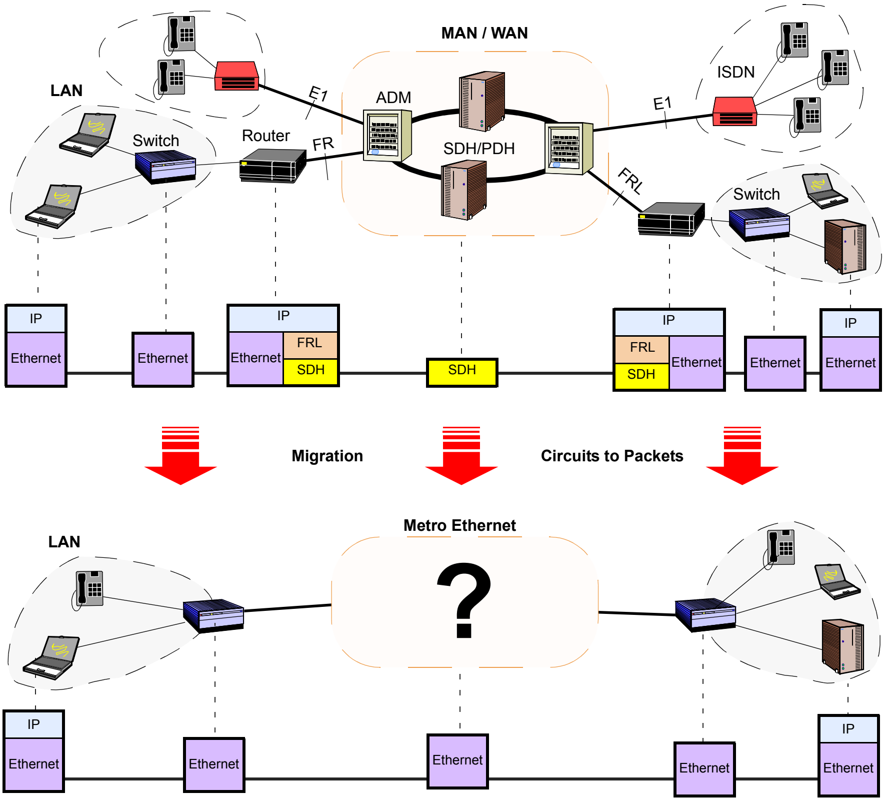


Figure 10.2 Migration to end-to-end Ethernet

The interface between the customer premises equipment and the service-provider facilities is called User-to-Network Interface (UNI). The fact that Ethernet is being offered as a service interface makes the definition of the Ethernet UNI very important. In fact, this is one of the main points addressed by standardization organizations. The deployment plans for the UNI include three phases:

1. UNI Type 1 focuses on the Ethernet users of the existing IEEE Ethernet physical and MAC layers.
2. UNI Type 2 requires static service discovery functionality with auto-discovery and OAM capabilities.
3. UNI Type 3 requires a dynamic connection setup such that *Ethernet Virtual Connections* (EVC) can be set up and / or modified from the customer UNI equipment.

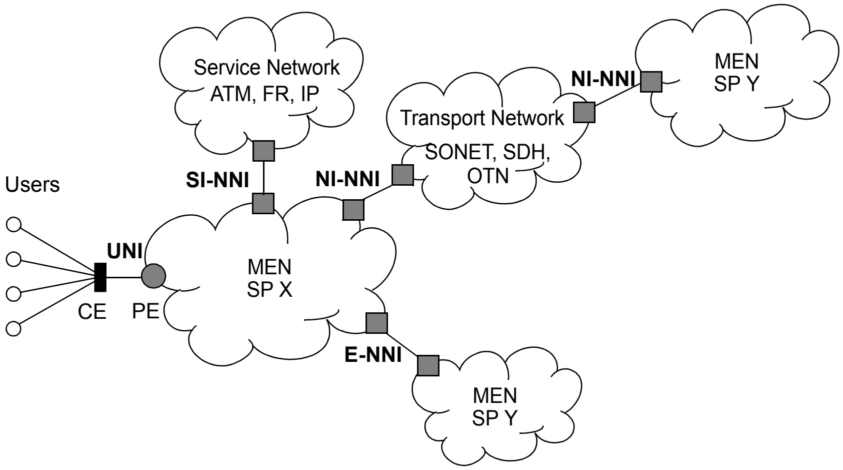


Figure 10.3 UNI and NNI in the MEN

The customer premises equipment that enables access to the MEN can be a router or a switch. This equipment is usually called Customer Edge (CE) equipment. The service provider equipment connected to the UNI, known as Provider Edge (PE), is a switch but deployments with routers are possible as well.

Many other interfaces are still to be defined, including the *Network-to-Network Interface* (NNI) for MEN inter-networking (see Figure 10.3). The network elements of the same MEN are connected by *Internal NNIs* (I-NNI). Two autonomous MENs are connected at an *External NNI* (E-NNI). The inter-networking to a transport network based on SDH, or *Optical Transport Network* (OTN), is done at the *Network Inter-Networking NNI* (NI-NNI). Finally, the connection to a different layer-2 network is established at the *Services Inter-Networking NNI* (SI-NNI).

10.1.2 Ethernet Virtual Connections

An *Ethernet Virtual Connection* (EVC) is defined as an association of two or more UNIs. A point-to-point EVC is limited to two UNIs, but a multipoint-to-multipoint EVC can have two or more UNIs that can be dynamically added or removed.

An EVC can be compared with the *Virtual Circuits* (VC) used by FR and ATM – however, the EVC has multipoint capabilities, whilst VCs are strictly point-to-point. This feature makes it possible to emulate the multicast nature of Ethernet. An EVC facilitates the transmission of frames between UNIs, but also prevents the transmission of information outside the EVC.

Origin and destination MAC addresses and frame contents remain unchanged in the EVC, which is a major difference compared to routed networks where MAC addresses are modified at each Ethernet segment.

10.1.3 Multiplexing and Bundling

An Ethernet port can support several EVCs simultaneously. This feature, called service multiplexing, improves port utilization by lowering the number of ports per switch. It also makes service activation more simple (see Figure 10.4). Service multiplexing is achieved by using the IEEE 802.1Q *Virtual LAN* (VLAN) ID as a connection identifier.

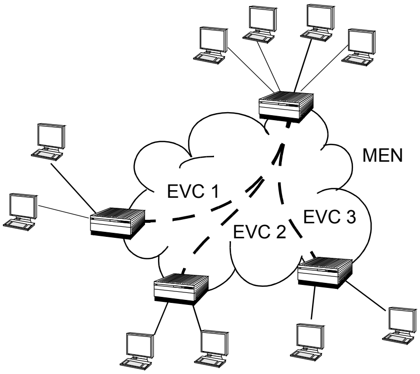


Figure 10.4 EVC Service multiplexing in a single port

Service multiplexing makes it possible to provide new services without installing new cabling or nodes. This, consequently, reduces capital expenditure.

Bundling occurs when more than one subscriber's VLAN ID is mapped to the same EVC. Bundling is useful when the VLAN tagging scheme must be preserved across the MEN when remote branch offices are going to be connected. A special case of bundling occurs when every VLAN ID is mapped to a single EVC. This is called *all-to-one bundling*.

10.1.4 Generic Service Types

Currently, the MEF has defined two generic service types: *Ethernet Line* (E-Line) and *Ethernet LAN* (E-LAN).

10.1.4.1 E-Line Service Type

The *E-Line service* is a point-to-point EVC with attributes such as *Quality of Service* (QoS) parameters, VLAN tag support, and transparency to layer-2 protocols (see Figure 10.5). The E-Line service can be compared, in some way, with *Permanent VCs* (PVCs) of FR or ATM, but E-Line is more scalable and has more service options.

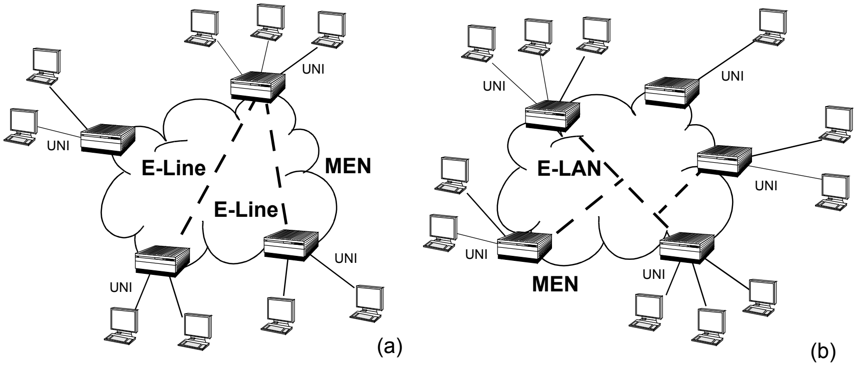


Figure 10.5 (a) The E-Line is understood as a point-to-point virtual circuit
(b) The E-LAN service is multipoint to multipoint

An E-Line service type can be a just simple Ethernet point-to-point with best effort connection, but it can also be a sophisticated TDM private line emulation.

10.1.4.2 E-LAN Service Type

The *E-LAN service* is an important new feature of Carrier-Class Ethernet. It provides a multipoint-to-multipoint data connection (see Figure 10.5). UNIs are allowed to be connected or disconnected from the E-LAN dynamically. The data sent from one UNI is sent to all other UNIs of the same E-LAN in the same way as happens in a classical Ethernet LAN. The E-LAN service offers many advantages over FR and ATM hub-and-spoke architectures that depend on various point-to-point PVCs to implement multicast communications.

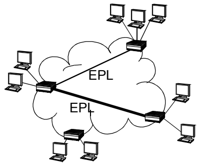
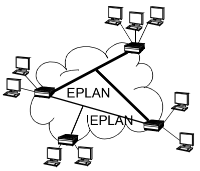
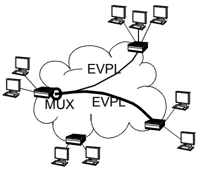
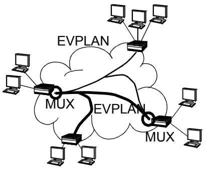
The E-LAN can be offered simply as a best-effort service type, but it can also provide a specific QoS. Every UNI is allowed to have its own bandwidth profile. This could be useful when several branch offices are connected to one central office. In this case the *Committed Information Rate* (CIR) in the UNI for every branch office could be 10 Mbit/s, and 100 Mbit/s for the central office.

10.1.5 Connectivity Services

An Ethernet service arises when a generic service type (E-Line or E-LAN) is offered with particular EVC and UNI features. When a service is port-based – that is, one single service per port is provided at the UNI, it is called *Ethernet Private Line* (EPL) or *Ethernet Private LAN* (EPLAN), depending on if it is point-to-point or multipoint-to-multipoint. Multiplexed services are called virtual. An *Ethernet Virtual Private Line* (EVPL) service and an *Ethernet Virtual Private LAN* (EVPLAN) service can be defined (see Table 10.1).

From the point of view of the customer, the main differences between virtual and non-virtual services are that EPLs and EPLANs provide better frame transparency, and they are subject to more demanding *Service Level Agreement* (SLA) margins than EVPLs and EVPLANs.

Table 10.1
Ethernet Connectivity Services

EVC to UNI Relationship	*Generic Etherservice Type*	
	E-Line - Point to point - Best-effort or guaranteed QoS - Optional multiplexing and bundling	E-LAN - Multipoint to multipoint - Best effort or guaranteed QoS - Optional multiplexing and bundling
Port-Based Service - No Service Multiplexing - Dedicated Bandwidth	Ethernet Private Line (EPL) 	Ethernet Private LAN (EPLAN) 
VLAN-Based Service - Service Multiplexing - Shared Bandwidth	Ethernet Virtual Private Line (EVPL) 	Ethernet Virtual Private LAN (EVPLAN) 

The meaning of multiplexed services in the case of EVPLs and EVPLANs needs to be further explained. For example, several E-Line service types may be multiplexed in different SDH timeslots and be still considered EPLs. This is because the *Time Division Multiplexing* (TDM) resource-sharing technique of SDH makes it possible to divide the available bandwidth in such a way that congestion in some timeslots does not affect other timeslots. This way, it is possible to maintain the strong SLA margins typical of EPLs and EPLANs in those timeslots that are not affected by congestion.

EVPLs and EVPLANs are statistically multiplexed services. They make use of service multiplexing (see Paragraph 10.1.3), and thus VLAN IDs are used as EVC identifiers at the UNI.

10.1.5.1 Ethernet Private Lines

The *Ethernet Private Line* (EPL) service is a point-to-point Ethernet service that provides high frame transparency, and it is usually subject to strong SLAs. It can be considered as the Ethernet equivalent of a private line, but it offers the benefit of an Ethernet interface to the customer.

The EPLs make use of all-to-one bundling and subscriber VLAN tag transparency. This allows the customer to easily extend the VLAN architecture between sites at both ends of the MAN/WAN connection. Frame transparency enables typical layer-2 protocols, such as IEEE 802.1q *Spanning Tree Protocol* (STP), to be tunneled through the MAN/WAN.

EPLs are sometimes delivered over dedicated lines, but they can be supplied by means of layer-1 (TDM or lambdas) or layer-2 (MPLS, ATM, FR) multiplexed circuits. Some service providers want to emphasize this, and they talk about dedicated EPLs, if dedicated lines or layer-1 multiplexed circuits are used to deliver the service, or shared EPLs if layer-2 multiplexing is used.

EPLs are the most extended Metro Ethernet services today. They are best suited for critical, real-time applications.

10.1.5.2 Ethernet Virtual Private Lines

The *Ethernet Virtual Private Line* (EVPL) is a point-to-point Ethernet service similar to the EPL, except that service multiplexing is allowed, and it can be opaque to certain types of frames. For example, STP frames can be dropped by the network-side UNI.

Shared resources make it difficult for the EVPL to meet SLAs as precise as those of EPLs. The EVPL is similar to the FR or ATM PVCs. The VLAN ID for EVPLs is the equivalent of the FR *Data Link Connection Identifier* (DLCI) or the ATM *Virtual Circuit Identifier* (VCI) / *Virtual Path Identifier* (VPI).

One application of EVPLs could be a high-performance ISP-to-customer connection.

10.1.5.3 Ethernet Private LANs

Ethernet Private LANs (EPLAN) are multipoint-to-multipoint dedicated Carrier-Class Ethernet services. The EPLAN service is similar to the classic LAN Ethernet service, but over a MAN or a WAN. It is a dedicated service in the sense that Ethernet traffic belonging to different customers is not mixed within the service-provider network.

Ethernet frames reach their destination thanks to the MAC switching supported by the service-provider network. Broadcasting, as well as multicasting are supported.

EPLAN services make use of all-to-one bundling and subscriber VLAN tag transparency to support the customer's VLAN architecture. Frame transparency is implemented in EPLANs to support LAN protocols across different sites.

10.1.5.4 Ethernet Virtual Private LANs

The *Ethernet Virtual Private LAN* (EVPLAN) is similar to the EPLAN, but EVPLANs are supported by a shared SP architecture instead of a dedicated one. The EVPLAN also has some common points with the EVPL. For example, the VLAN tag is used for service multiplexing, and EVPLANs could be opaque to some LAN protocols, such as the STP.

Both EPLAN and EVPLAN will probably be the most important Carrier-Class Ethernet services in the future. They have many attractive features. The same technology, Ethernet, is used in LAN, MAN and WAN environments. One connection to the service-provider network per site is enough, and EPLAN and EVPLAN offer an interesting alternative to today's layer-3 VPNs. EPLANs and EVPLANs enable the customers to deploy their own IP routers on top of the layer-2 Ethernet VPN.

10.2 End-to-End Ethernet

Today's installations use Ethernet on LANs to connect servers and workstations. Data applications and Internet services use WANs to get or to provide access to / from remote sites by means of leased lines, PDH / SDH TDM circuits and ATM / FR PVCs. Routers are the intermediate devices that using IP as a common language can also talk to the LAN and WAN protocols. This has been a very popular solution, but it is not a real end-to-end Ethernet service. This means that MAC frames “die” as soon as IP packets enter on the PDH/SDH domain, and they are created again when they reach the far-end.

This option, which is now often considered as legacy, has been the most popular networking data solution. During the past couple of decades, routing technologies have formed flexible and distributed layer-3 VPNs. Since Ethernet is present in both LANs, why not use Ethernet across the WAN as well?

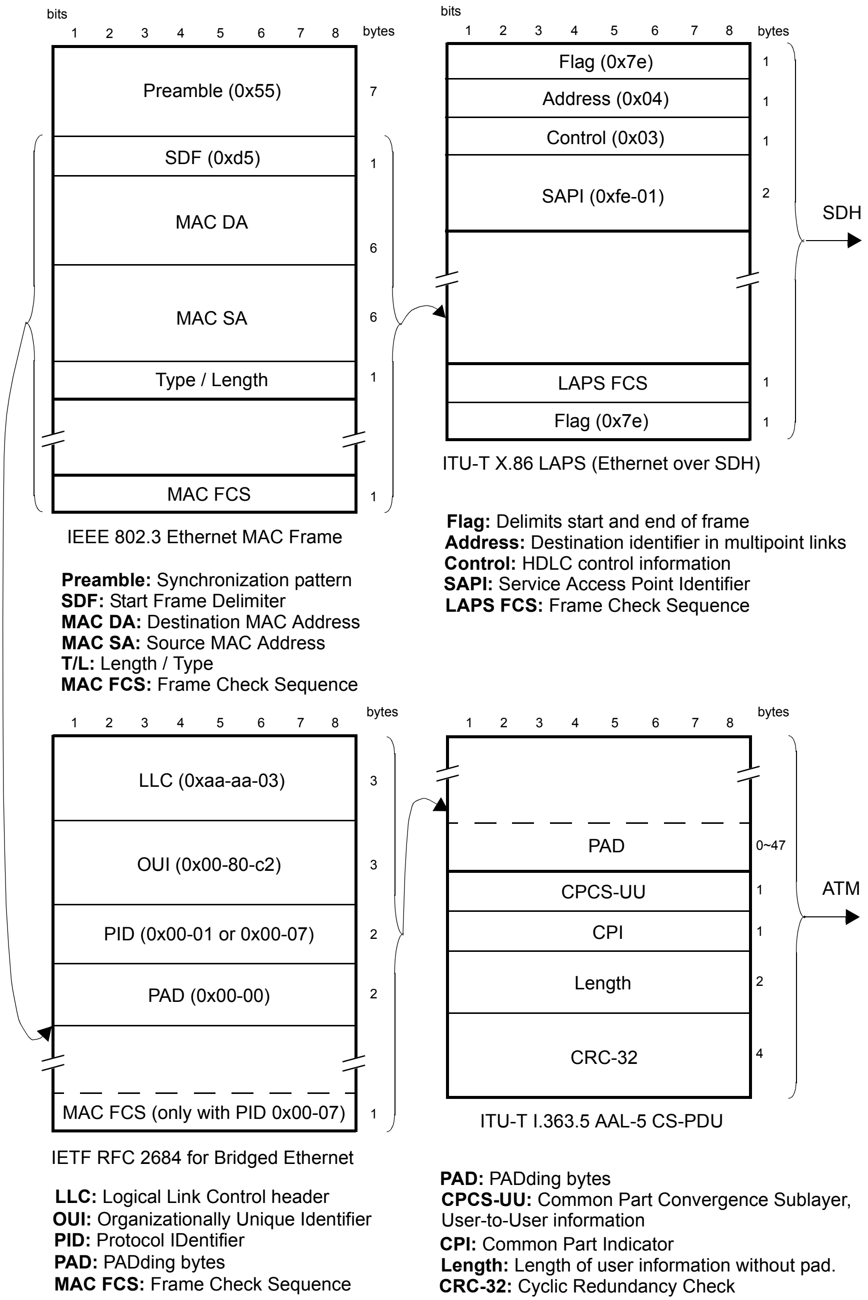
The first approaches for extending Ethernet over a WAN are based on mixing Ethernet with legacy technologies, for example Ethernet over ATM, as defined in the IETF standard RFC-2684, or Ethernet over SDH by means of the *Link Access Procedure - SDH* (LAPS) as per ITU-T Recommendation X.86 (see Figure 10.6).

- The LAPS is a genuine Ethernet solution that provides bit rate adaptation and frame delineation. It offers LAN connectivity, allowing switches and hubs to interface directly with classic SDH. But it uses a byte-stuffing technique that makes the length of the frames data-dependent. This solution tunnels the Ethernet frames over SDH TDM timeslots called Virtual Containers. The Ethernet MAC frames remain passive within the network, and therefore this solution is only useful for simple solutions, such as point-to point dedicated circuits.
- The solution based on Ethernet over ATM is more flexible and attractive for service providers, because it allows to set up point-to-point switched circuits based on ATM PVCs. With this solution, Ethernet frames are tunneled across the ATM network. Switching is based on ATM VPI / VCI fields. The main problems of this architecture are high cost and low efficiency, combined with the poor scalability of ATM.

The proposed alternatives, generically known as Carrier-Class Ethernet, replace ATM, FR or other layer-2 switching by Ethernet bridging based on MAC addresses (see Figure 10.7). Several architectures can fulfil the requirements, including dark fiber, DWDM / CWDM, NG-SDH. In principle, all of these architectures are able to support Carrier Ethernet services such as E-Line and E-LAN – however, some are more appropriate than others.

10.2.1 Optical Ethernet

Ethernet can now be used in metropolitan networks due to the recent standardization of new long-range, high-bandwidth Ethernet interfaces. It can be said that Ethernet bandwidths and ranges are now of the same order as the bandwidths and ranges provided by classical WAN technologies.



Preamble: Synchronization pattern
SDF: Start Frame Delimiter
MAC DA: Destination MAC Address
MAC SA: Source MAC Address
T/L: Length / Type
MAC FCS: Frame Check Sequence

Flag: Delimits start and end of frame
Address: Destination identifier in multipoint links
Control: HDLC control information
SAPI: Service Access Point Identifier
LAPS FCS: Frame Check Sequence

LLC: Logical Link Control header
OUI: Organizationally Unique Identifier
PID: Protocol IDentifier
PAD: PADding bytes
MAC FCS: Frame Check Sequence

PAD: PADding bytes
CPCS-UU: Common Part Convergence Sublayer, User-to-User information
CPI: Common Part Indicator
Length: Length of user information without pad.
CRC-32: Cyclic Redundancy Check

Figure 10.6 Legacy encapsulations for transporting Ethernet over SDH or ATM networks. The SDH solution makes use of the of the LAPS encapsulation. The ATM mapping uses RFC-2684 and AAL-5 encapsulations.

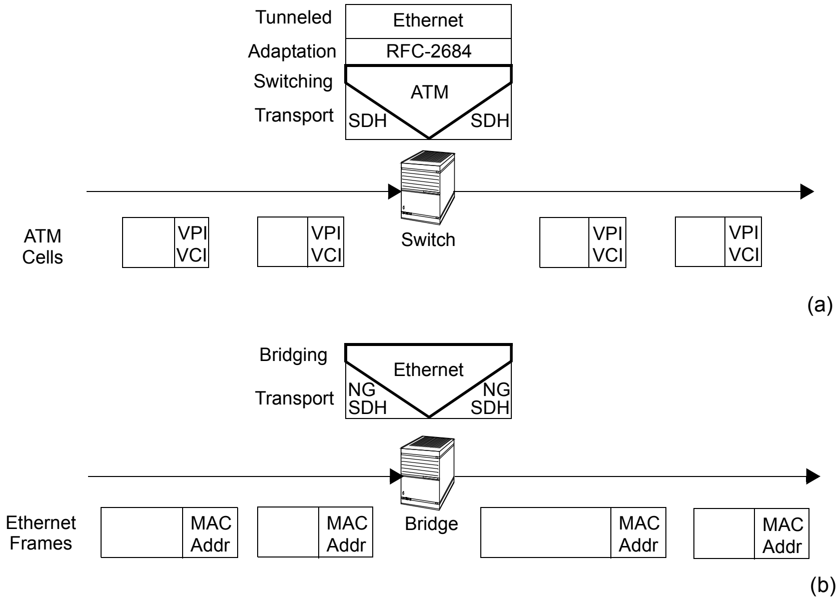


Figure 10.7 How NG-SDH raises the importance of Ethernet in the MAN / WAN. (a) Ethernet traffic is passively transported like any other user data. The ATM layer, specific for the WAN, is used for switching traffic. (b) The ATM layer disappears and the Ethernet layer becomes active. Traffic is now guided to its destination by means of Ethernet bridging.

MENs based on optical Ethernet are typical of early implementations. They are built by means of standard IEEE interfaces over dark optical fiber. They are therefore pure Ethernet networks. Multiple homing, link aggregation and VLAN tags can be used in order to increase resilience, bandwidth and traffic segregation. Interworking with the legacy SDH network can be achieved with the help of the WAN Interface Subsystem (WIS). The WIS is part of the WAN PHY specification for 10-Gigabit Ethernet. It provides multigigabit connectivity across SDH and WDM networks as an alternative to the LAN PHY for native-format networks.

With this simple solution, a competitive operator can take advantage of packet switching, multipoint-to-multipoint applications and quick service roll-out.This option is a cost-effective in those areas where spare dark fiber is available and tree topologies are likely. Despite of its simplicity, pure Ethernet solutions for MEN have big scalability problems. Furthermore, they suffer from insufficient QoS, OAM, and resilience mechanisms (see Paragraph 10.3).

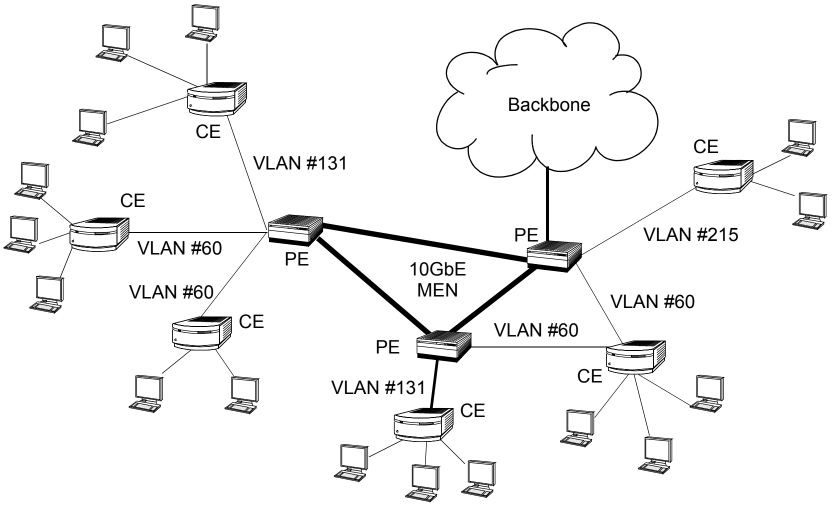


Figure 10.8 Optical Ethernet Carrier network. The trunk MEN links are implemented with optical 10-Gigabit Ethernet. Access links can be based on 1-Gigabit Ethernet or *Ethernet in the First Mile* (EFM), depending on the bandwidth requirements of every placement. Segregation of traffic from different subscribers or work groups is done by using VLAN tagging.

Optical Ethernet is often the architecture implemented by new operators to compete with the incumbent ones (see Figure 10.8). This kind of solution is still practical in metropolitan environments with a small number of connected subscribers or subsidiaries. Specifically, the pure Ethernet over dark fiber approach is discouraged for operators who want to provide services to a large number of residential and Small Office/Home Office (SOHO) customers.

10.2.2 Ethernet over WDM

The transport capability of the existing fiber can be multiplied by 16 or more if *Wavelength-Division Multiplexing* (WDM) is used. The resulting wavelengths are distributed to legacy and new technologies such Ethernet (see Figure 10.14) that will get individual lambdas while sharing fiber optics. WDM is a good option for core networks serving very high bandwidth demands from applications like triple play, remote backups or hard disk mirroring. However, cost can be a limiting factor.

One of the inconveniences of this approach is the need to keep track of different and probably incompatible management platforms: one for Ethernet, another one for lambdas carrying SDH or other TDM technologies, and finally a third one for WDM. That makes OAM, traffic engineering and maintenance difficult.

10.2.3 Ethernet over SDH

Solutions for transporting Ethernet over SDH based on the *Generic Framing Procedure* (GFP), Virtual Concatenation and the *Link Capacity Adjustment Scheme* (LCAS) are generically known as *Ethernet over SDH* (EoS). The idea behind EoS is to substitute the native Ethernet layer 1 by SDH. The Ethernet MAC layer remains untouched to guarantee as much compatibility as possible with the IEEE Ethernet. Due to this, EoS cannot be considered as a true Ethernet technology. However, it is of great importance, because SDH is the *de facto* standard for transport networks. EoS makes it possible to reuse the existing infrastructure by taking advantage of the best of the SDH world, including resilience, long range and extended OAM capabilities.

NG-SDH unifies circuit and packet services under a unique architecture, providing Ethernet with a reliable infrastructure very rich in OAM functions (see Figure 10.9).

The three new elements that have made this migration possible are:

1. *Generic Framing Procedure* (GFP), as specified in Recommendation G.7041, is an encapsulation procedure for transporting packetized data over SDH. In principal, GFP performs bit rate adaptation and mapping into SDH circuits.
2. *Virtual Concatenation* (VCAT), as specified in Recommendation G.707, creates channels of customized bandwidth sizes rather than the fixed bandwidth provision of classic SDH, making transport and bandwidth provision more flexible and efficient.
3. *Link Capacity Adjustment Scheme* (LCAS), as specified in Recommendation G.7042, can modify the bandwidth of the VCAT channels dynamically, by adding or removing bandwidth elements of the channels, also known as members.

Ethernet traffic can be encapsulated in two modes:

1. *Transparent GFP* (GFP-T) is equivalent to a leased line with the bandwidth of the Ethernet bit rate. No delays, but expensive.
2. *Framed GFP* (GFP-F) is more efficient, because it removes interframe gaps and unnecessary frame fields. It also allows bandwidth sharing among several traffic flows. With GFP-F, service providers benefit from the statistical multiplexing gain, although subscribers may receive reduced performance when compared to GFP-T. This is due to the use of queues that increase end-to-end delay. Differentiated traffic profiles can be offered to customer signals (see Figure 10.10).

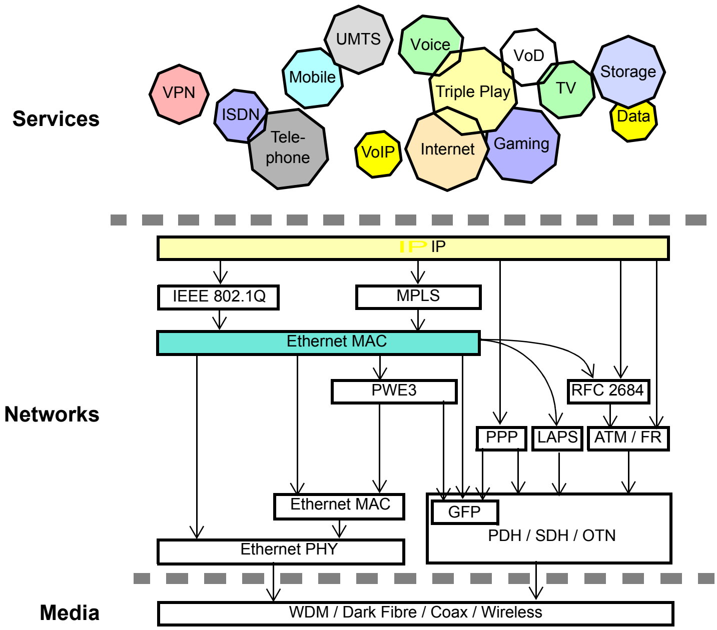
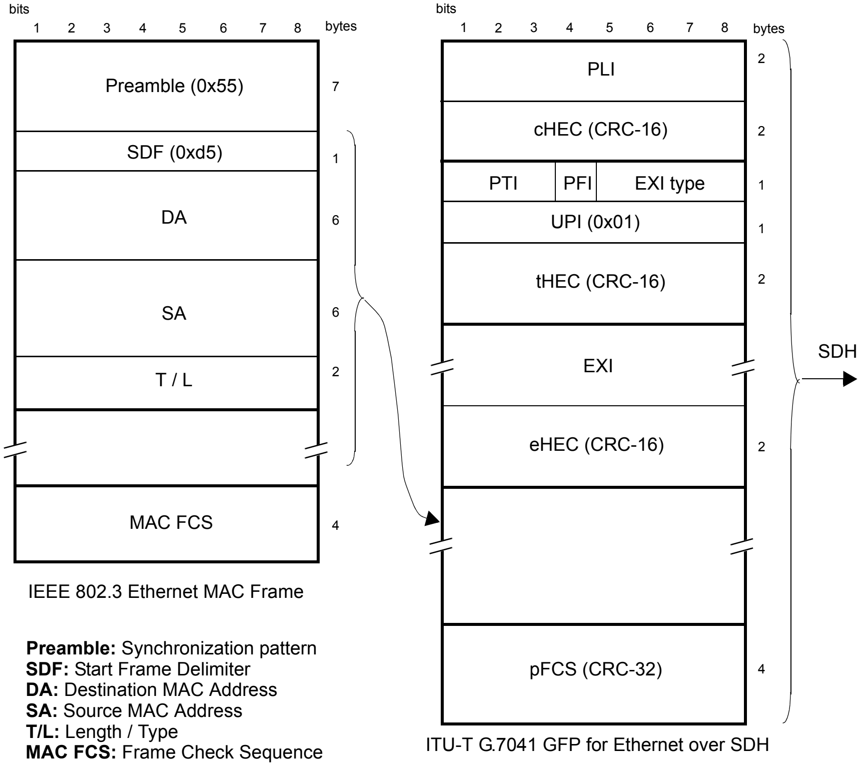


Figure 10.9 Transporting Ethernet and IP over packet- or circuit-switched infrastructures

Compared to ATM, the GFP-F encapsulation has at least three critical advantages:

1. It adds very little overhead to the traffic stream. ATM adds 5 overhead bytes for every 53 delivered bytes plus AAL overhead.
2. It carries payloads with variable length, as opposed to ATM that can only carry 48-byte payloads. This makes it necessary to split long packets into small pieces before they are mapped in ATM.
3. It has not been designed as a complete networking layer like ATM – it is just an encapsulation. Specifically, it does not contain VPI / VCI or other equivalent fields for switching traffic. Switching is left to the upper layer, usually Ethernet.

Like LAPS, GFP can be used for tunneling of Ethernet traffic over an SDH path, but the importance of this new mapping is that it allows Ethernet traffic to be active within the WAN. With the help of GFP, SDH network elements are able to bridge MAC frames like any other switch based on the Ethernet physical layer. The features



IEEE 802.3 Ethernet MAC Frame

ITU-T G.7041 GFP for Ethernet over SDH

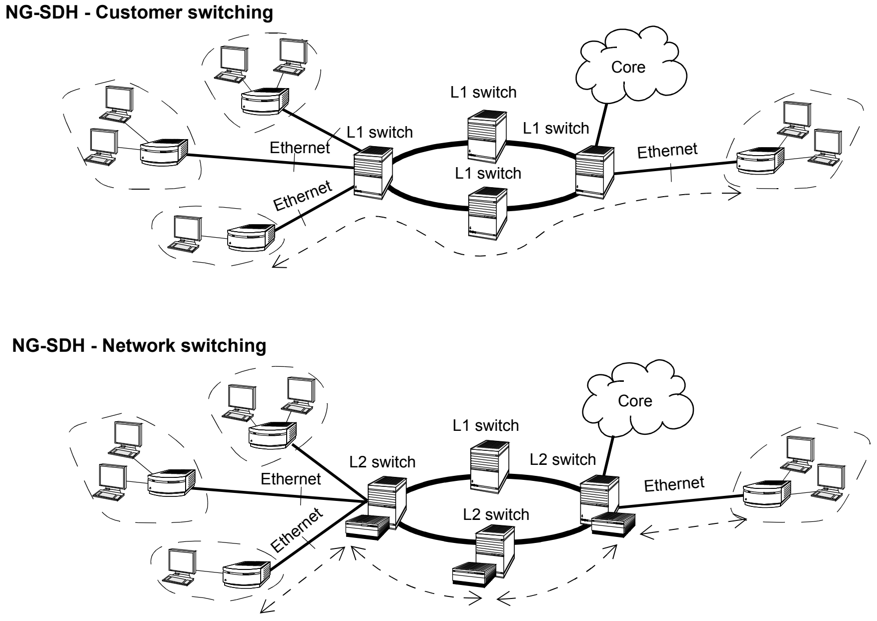
Preamble: Synchronization pattern
SDF: Start Frame Delimiter
DA: Destination MAC Address
SA: Source MAC Address
T/L: Length / Type
MAC FCS: Frame Check Sequence

PLI: PDU Length Indicator
cHEC: core HEC protection
PTI: Payload Type Identifier
PFI: Payload FCS Indicator
EXI type: Specifies the EXI header format
UPI: User Payload Identifier
tHEC: type HEC protection
EXI: Extension Header Identifier
pFCS: payload Frame Check Sequence

Figure 10.10 The GFP-F mapping for Ethernet makes ATM unnecessary. Now there is no VPI / VCI to switch the traffic, but the Ethernet MAC addresses can be used for similar purposes.

of SDH MAC switches include MAC address learning and flooding of frames with unknown destination MAC address. In a few words, SDH MAC switches enable us to emulate an Ethernet LAN over an SDH network (see Figure 10.11).

Deploying Ethernet in MAN / WAN environments makes it necessary to develop new types SDH *Add / Drop Multiplexers* (ADMs) and *Digital Cross-Connects* (DXC) with layer-2 bridging capabilities (see Figure 10.12):



NG-SDH - Customer switching

NG-SDH - Network switching

Figure 10.11 Ethernet over NG-SDH. Depending on the requirements, two approaches are possible: a) Customer switching – simple; the transport network is just a link between the customer switches. b) Network switching – more flexible; one step forward toward a more sophisticated service based on MPLS.

- Enhanced ADMs are like a traditional ADM, but they include Ethernet interfaces to enable access to new services, and TDM interfaces for legacy services. Many of these network elements add Ethernet bridging capabilities, and some support MPLS and *Resilient Packet Ring* (RPR). New services benefit from the advantages of NG-SDH. New and legacy services are segregated in different SDH TDM timeslots.
- Packet ADMs have a configuration similar to enhanced ADMs: They include TDM and packet interfaces but packet ADM offers common packet-based management for both new and legacy services. The TDM tributaries are converted into packets before being forwarded to the network. *Circuit Emulation over Packet* (CEP) features are needed. MPLS is likely to be the technology in charge of multiplexing new and legacy services together in packet ADMs, due to the flexibility given by MPLS connections known as *Label-Switched Paths* (LSP). Packet ADMs provide the same advantages as enhanced ADMs, but additionally, the network operator can benefit from increased efficiency and simplified management due to a unified switching paradigm.

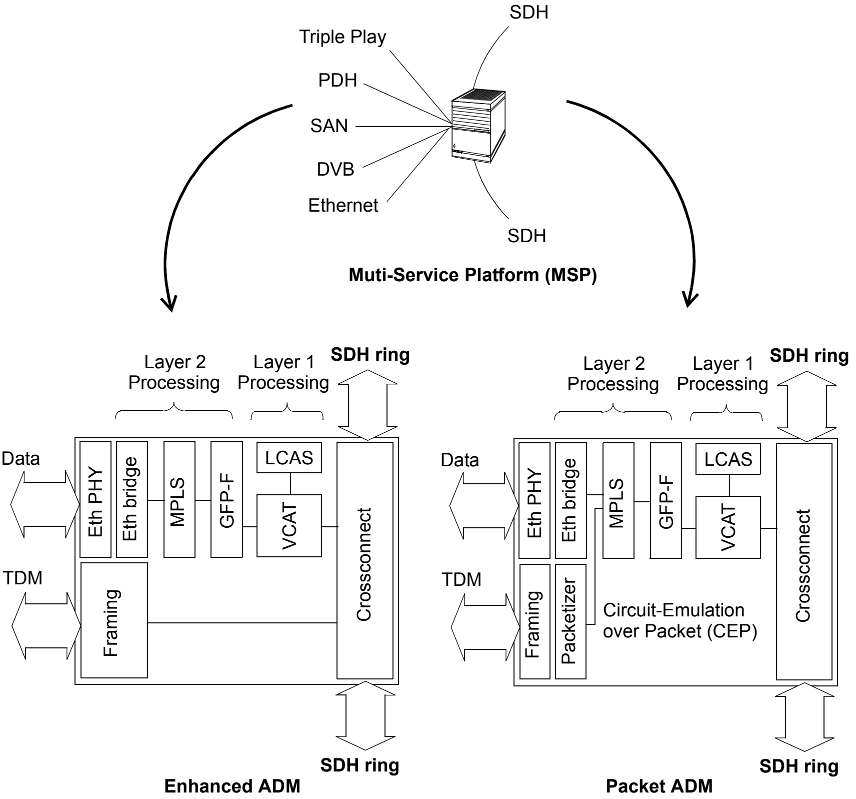


Figure 10.12 New SDH network elements. The Enhanced ADM offers packet and TDM interfaces in the same network element. Packet ADMs offer the same, but over an unified packet-based switching paradigm for all tributaries.

EoS is the technology preferred by incumbent operators, as they already have a large basis of SDH equipment in use. On the other hand, new operators generally prefer Carrier Ethernet directly implemented over optical layers.

10.3 Limitations of Bridged Networks

Metro Ethernet architectures, based only on native Ethernet switches and any combination of dark fiber, SDH and WDM, are like a big LAN – they have the same advantages and similar disadvantages. We know that in low traffic conditions and with a limited number of hosts, this network works very well. But as soon as the installation begins to grow, aspects like scalability, quality of service, topologies and protection tend to fall down.

Let us examine the drawbacks in more detail.

Table 10.2
Alternatives for providing Carrier Ethernet services

	Fiber	*Fiber/WDM + MPLS*	*CWDM or DWDM*	*NG-SDH*	*NG-SDH + PWE3*	*Classic SDH (VC / LAPS)*
				Low latency, low jitter, high resiliency, good OAM		
Service	E-Line E-LAN	E-Line E-LAN if VPLS	E-Line E-LAN if VPLS	E-Line	VPWS: E-Line VPLS: E-LAN	E-Line
Bandwidth	High	High	High	High	High	Inefficient
Cost	Medium-low	Medium-low	Very expensive	Medium, if SDH exits	Medium, if SDH exits	Low
QoS	Best effort	Best effort Carrier if MPLS	Carrier	Carrier	Carrier	Not differentiated
SLA	EIR, PIR	CIR, EIR	Guaranteed	GFP-F: EIR>CIR GFP-T: EIR=CIR	EIR, CIR	EIR=CIR
Legacy TDM	No	Yes	No	Yes	Yes	Reliable Predictable
Statistical	Yes	Yes	Yes	Yes	Yes	No
Latency	Medium	Medium	Very Low	Low	Very Low	Very Low
Jitter	High	High	Low	Very Low	Very Low	Very Low
Topology	Meshed	Meshed	Meshed	Ring Linear	Ring Linear	Ring Linear
Network	Access Metro	Access Metro	Core Data Center	Metro Core	Metro Core	Access Metro
Good for	VPN Triple Play if MPLS	Internet Triple Play if MPLS	Any including Storage	Data, VoIP, Video, Connectivity	Time sensitive applications	Superseded
Resilience	STP ~10 s RSTP ~1 s RPR 50 ms	STP ~10 s RSTP ~1 s RPR 50 ms	OTN 50ms	APS 50 ms LCAS 50 ms	Path protection APS 50 ms LCAS 50 ms	APS 50 ms
MPLS	Yes	Yes	Yes	Yes	VPLS and VPWS	No
Management	Distributed	Distributed	Distributed	Hierarchical & Distributed	Hierarchical & Distributed	Hierarchical

10.3.1 Scalability

Ethernet switches use promiscuous broadcasting to learn addresses constantly (IEEE 802.1D). When a request to forward a frame to an unknown address arrives, the switch has to flood the frame to all the ports, waiting for a response to know where the address is. This way is not very efficient, nor secure.

MAC addresses are not hierarchical, and the switching table does not scale very well, slowing down the performance of switches when there is a large number of client hosts (this is known as the MAC switching table explosion problem). Furthermore, all the switches in the network have to constantly learn the MAC addresses of new stations connected to the network.

VLANs (IEEE 802.1Q) offer an easy solution to some of the problems mentioned before. A switch can be divided into smaller virtual switches, each of them belonging to a different VLAN. VLANs are used to split one big broadcast domain into several smaller domains, thus improving security and reducing broadcast traffic.

Another advantage of 802.1Q VLANs is the ability to offer QoS by means of the three 802.1p bits (VLAN CoS bits).

However, the number of VLAN identifiers is limited to 4,096. This limits the number of subscribers a service provider can have in the same network. Furthermore, subscribers may have their own VLAN structure, and it is desirable to support the customer and the provider VLAN structure simultaneously.

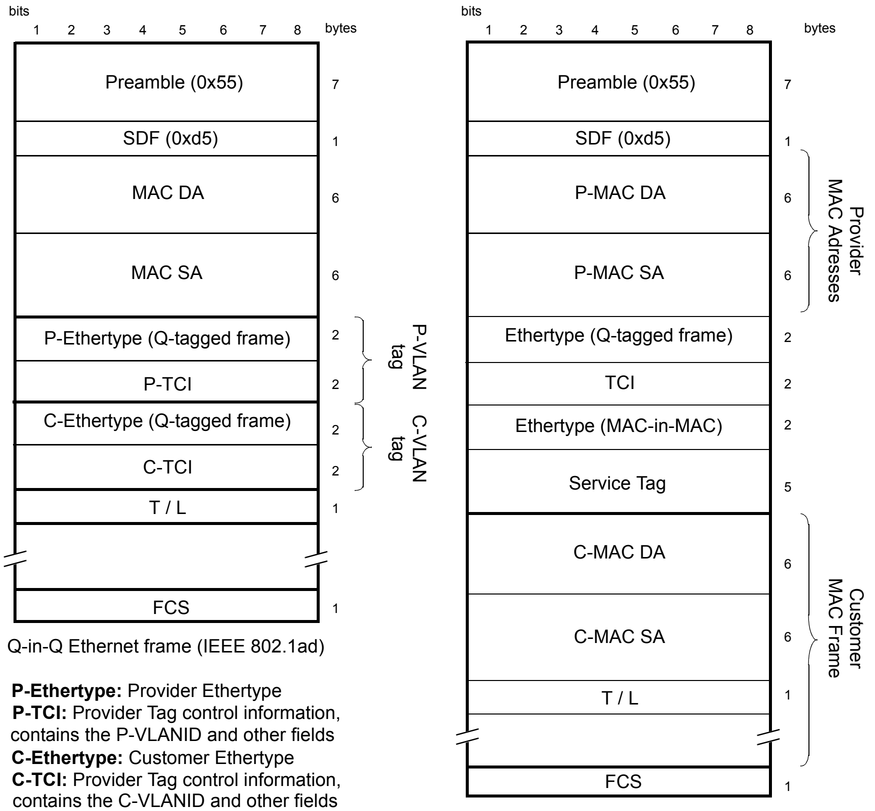
A solution is to stack VLAN tags (Q-in-Q solution) to obtain a larger number of VLAN identifiers as is specified in the IEEE 802.1ad standard for provider Ethernet bridges. *MAC Address Stacking* (MAS) is a more powerful solution, designed to deal with MAC switching table explosions (see Figure 10.13). The customer equipment is not supposed to understand the Q-in-Q and MAS frame formats. This means that the service provider VLAN labels and MAC addresses are added by the first hop and removed by the last hop in the carrier network.

10.3.2 Protection

The *Spanning Tree Protocol* (STP) and the *Rapid Spanning Tree Protocol* (RSTP) work well enough in LAN and also in data services. But they do not meet the challenge when it comes to the mass rollout of IP services, or the 50 ms restoration time paradigm of carrier-class services.

10.3.3 Topologies

All topologies are possible, but some may become inefficient due to the operation of the STP or any of its variations. This is the case with rings which are used only partially; they are transformed into trees, and unused links are left just for protection. The IEEE 802.17 for *Resilient Packet Rings* (RPR) was designed to solve this problem, but it has not gained market acceptance, probably because it is rather expensive and not flexible enough, as it requires the same bit rate for all nodes.



Q-in-Q Ethernet frame (IEEE 802.1ad)

P-Ethertype: Provider Ethertype
P-TCI: Provider Tag control information, contains the P-VLANID and other fields
C-Ethertype: Customer Ethertype
C-TCI: Provider Tag control information, contains the C-VLANID and other fields

MAC-in-MAC Ethernet frame (IEEE 802.1ah)

P-MAC SA: Provider source MAC address
P-MAC DA: Provider destination MAC address
C-MAC SA: Customer source MAC address
C-MAC DA: Customer destination MAC address

Figure 10.13 New frame formats for provider Ethernet bridges. These frames offer a more scalable Ethernet for carrier networks.

Another problem is network demarcation, when the same technology is everywhere without a clear border between customer and provider installations. It is necessary to find a solution to questions like:

- Can the typical LAN protocols like STP, deployed by subscribers in their networks, influence the overall service-provider network operation?

- How will uncontrolled traffic generated by some subscribes affect the global MAN / WAN operation?
- How will the continuous connection and disconnection of stations in the subscriber network affect the service provider network?

These problems cannot be solved without important enhancements in the classic Ethernet technology, and installation of the necessary demarcation devices to isolate and filter the traffic between networks.

10.3.4 Quality of Service

Up to eight levels of priority can be set up to VLAN-tagged Ethernet frames (IEEE 802.1Q/p) to manage different traffic classes – however, native Ethernet is not really a QoS-enabled technology. In fact, Ethernet is unable to provide services with guaranteed QoS, because it doesn't have resource management and traffic engineering tools.

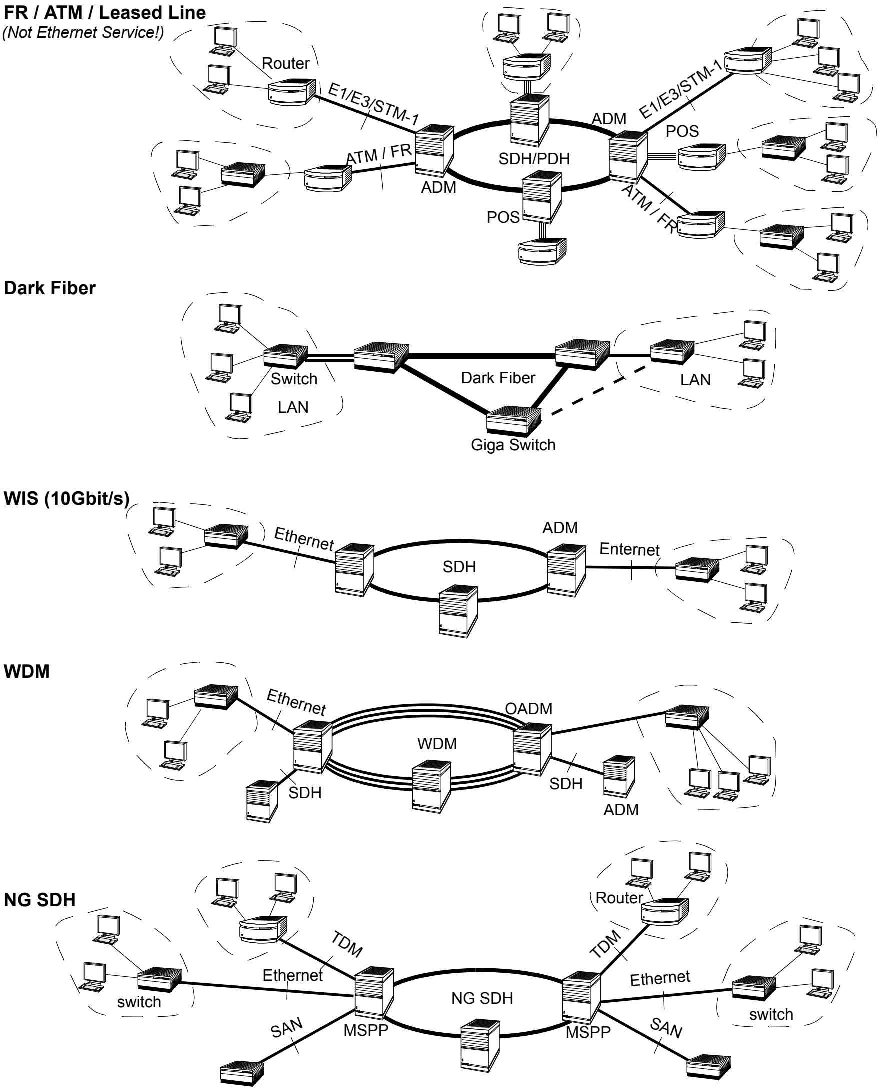
VLAN, and prioritization standards[1] were designed for LAN and do not scale well on WAN. Features like QoS, security, availability, and performance can be seriously damaged. That is why Metropolitan Operators of Ethernet networks must rely on other technologies like MPLS to overpass most of these native Ethernet limitations.

10.4 MULTI-PROTOCOL LABEL SWITCHING

Multi-Protocol Label Switching (MPLS) is a technology designed to speed up IP packet switching in routers by separating the functions of route selection and packet forwarding into two planes:

- *Control Plane*: This plane manages route learning and selection with the help of traditional routing protocols such as *Open Shortest Path First* (OSPF) or *Intermediate System - Intermediate System* (IS-IS). In a network element, the MPLS control plane maintains the routing table.
- *Forwarding Plane*: This plane switches IP packets, taking as a basis short labels prepended to them. To do this, the forwarding plane needs to maintain a switching table that associates each incoming labeled packet with an output port and a new label.

[1] IEEE 802.1Q for VLAN, IEEE 802.1Q/p for priority, IEEE 802.1D for bridging/switching.

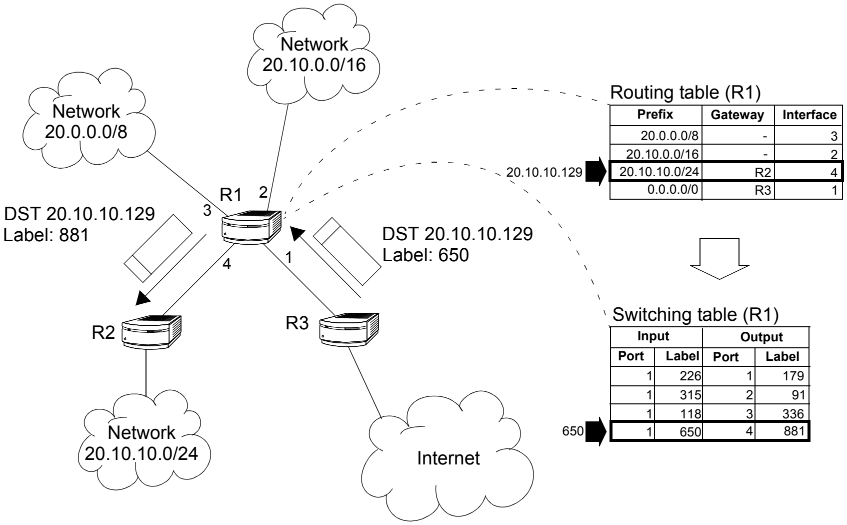


FR / ATM / L.Line *Not Ethernet Service!*	Dark Fiber	WDM	WIS (SDH framing)	LAPS / NG SDH	MPLS
IP	IP	IP	IP	IP	IP
FR / ATM / PPP	Ethernet MAC	Ethernet MAC	Ethernet MAC	Ethernet MAC	Ethernet MAC
PDH / SDH	Ethernet PHY	Ethernet PHY	WIS	Adaptation	MPLS
		WDM	10GbE PHY	SDH / SONET	Any PHY

Figure 10.14 Alternatives for providing Carrier Ethernet services

The traditional IP routers switch packets according to their routing table. This mechanism involves complex operations that slow down switching. Specifically, traditional routers must find the longest network address prefix in the routing table that matches the destination of every IP datagram entering the router.

On the other hand, MPLS routers, also known as *Label-Switched Routers* (LSR), use simple, fixed-length label forwarding instead of a variable-length IP network prefix for fast forwarding of packetized data (see Figure 10.15).



Routing table (R1)

Prefix	Gateway	Interface
20.0.0.0/8	-	3
20.10.0.0/16	-	2
20.10.10.0/24	R2	4
0.0.0.0/0	R3	1

Switching table (R1)

Input		Output	
Port	Label	Port	Label
1	226	1	179
1	315	2	91
1	118	3	336
1	650	4	881

Figure 10.15 Traditional routers have to perform complex operations to resolve the output interface of incoming packets. LSRs resolve the output interface with the help of a simple switching table.

MPLS enables the establishment of a special type of virtual circuits called *Label-Switched Paths* (LSP) in IP networks. Thanks to this feature, it is possible to implement resource management mechanisms for providing hard QoS on a per-LSP basis, or to deploy advanced traffic engineering tools that provide the operator with tight control over the path that follows every packet within the network. Both QoS provision and advanced traffic engineering are difficult, if not impossible to solve in traditional IP networks.

To sum up, the separation of two planes allows MPLS to combine the best of two worlds: the flexibility of the IP network to manage big and dynamic topologies automatically, and the efficiency of connection-oriented networks by using preestablished paths to route the traffic in order to reduce packet process on each node.

10.4.1 Labels

When Ethernet is used as the transport infrastructure, it is necessary to add an extra “shim” header between the IEEE 802.3 MAC frames and the IP header to carry the MPLS label. This MPLS header is very short (32 bits), and it has the following fields (see Figure 10.16):

- **Label (20 bits)**: This field contains the MPLS label used for switching traffic.
- **Exp (3 bits)**: This field contains the experimental bits. It was first thought that this field could carry the 3 Type-of-Service (ToS) bits defined for traffic differentiation in the IP version 4, but currently, the ToS field is being replaced by 6-bit *Differentiated Services Code Points* (DSCP). This means that only a partial mapping of all the possible DSCPs into the Exp bits is possible.
- **S (1 bit)**: This bit is used to stack MPLS headers. It is set to 0 to show that there is an inner label, otherwise it is set to 1. Label stacking is an important feature of MPLS, because it enables network operators to establish LSP hierarchies.
- **TTL (8 bits)**: This field contains a *Time To Live* value that is decremented by one unit every time the packet traverses an LSR. The packet is discarded if the value reaches 0.

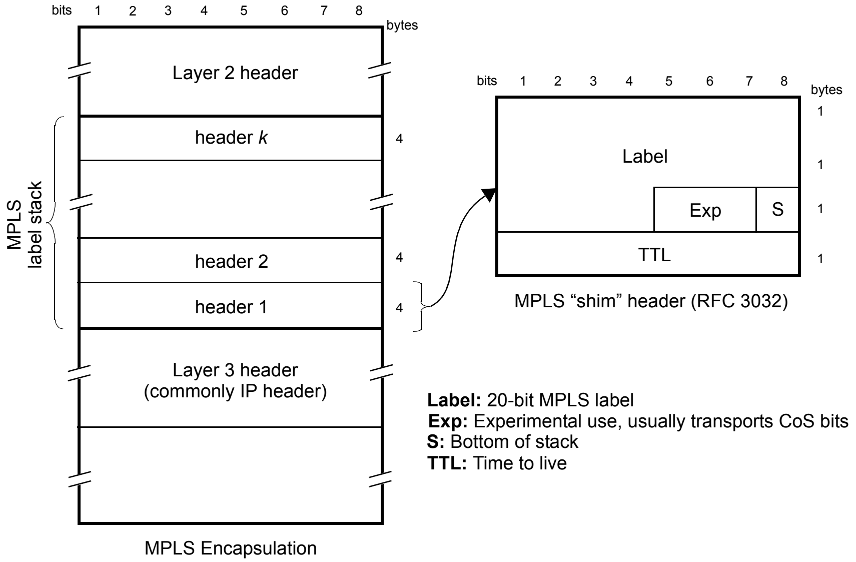


Figure 10.16 MPLS “shim” header format. The label is usually inserted between the layer-2 and layer-3 headers.

MPLS can be used in SDH transport infrastructures as well. IP routers with SDH interfaces can benefit from the advantages of MPLS like any other IP router. Since the MPLS header must be inserted between layer-2 and layer-3 headers, it is necessary to encapsulate MPLS-labeled frames into Ethernet MAC frames before they are mapped to SDH. However, recent ITU-T recommendations allow direct mapping of MPLS-labeled packets to GFP-F for transport across NG-SDH circuits. This is an important exception of the common frame labeling, because in this case labels are inserted between a layer-1 header (GFP-F) and a layer-3 header (IP). This new mapping improves efficiency of SDH LSRs by eliminating the need of a passive Ethernet encapsulation used only for adaptation (see Figure 10.17).

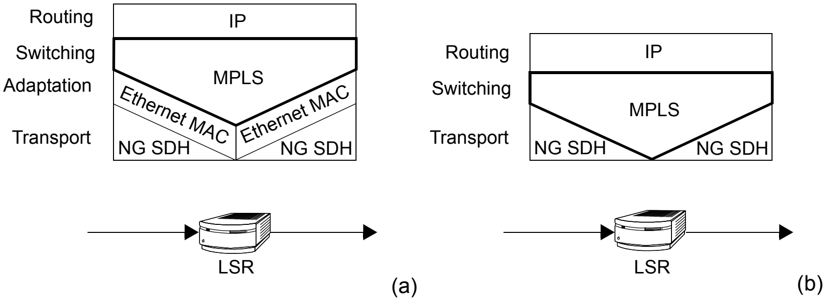


Figure 10.17 Protocol stacks of SDH LSRs. (a) Traditional protocol stack for an SDH LSR: the MPLS header is inserted between layer-2 (Ethernet MAC) and layer-3 (IP) headers. (b) Direct mapping of MPLS over SDH: The MPLS header is mapped between a layer-1 (GFP-F) overhead and layer-3 (IP) overhead without the need of a passive Ethernet encapsulation only used for adaptation.

The MPLS label is sometimes included in the "shim" header inserted between the layer-2 and layer-3 headers, but this is not always true. Almost any header field used for switching can be reinterpreted as an MPLS label. The FR 10-bit *Data Link Connection Identifier* (DLCI) field or the ATM *Virtual Path Identifier* (VPI) and *Virtual Circuit Identifier* (VCI) are two examples of this. The ATM VPI / VCI example is of special importance, because it allows a smooth transition from the ATM-based network core to an IP / MPLS core. An ATM switch can be used as an LSR with the help of relatively simple upgrade that will probably involve only new software.

MPLS is proving to be a technology with incredible flexibility. Timeslot numbers in TDM frames, or even wavelengths in WDM signals can be re-interpreted as MPLS labels as well. This approach opens the door to a new way of managing TDM / WDM networks. The MPLS-based management plane for TDM / WDM networks uses distributed IP routing, and at the same time it benefits from the powerful traffic engineering features of MPLS. This, in fact, forms a new technology and an active investigation field called Generalized MPLS (GMPLS).

10.4.2 MPLS Forwarding Plane

Whenever a packet enters an MPLS domain, the ingress router, known as ingress *Label Edge Router* (LER), inserts a header that contains a label that will be used by the LSR to route packets to their destination. When the packet reaches the edge where the egress router is, the label is dropped and the packet is delivered to its destination (see Figure 10.18). Only input labels are used for forwarding the packets within the network, while encapsulated addresses like IP or MAC are completely ignored.

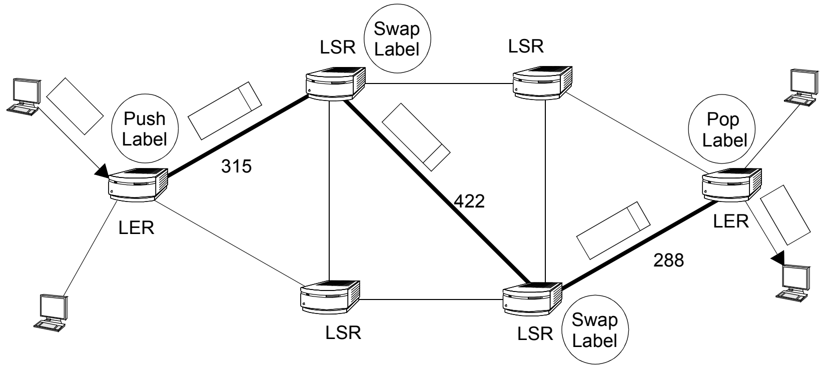


Figure 10.18 Label processing within an MPLS domain. A label is pushed by the ingressing LER, swapped by the intermediate LSR across the LSP, and popped by the egressing LER.

In typical applications, labels are chosen to force the IP packets to follow the same paths they would follow if they were switched with routing tables. This means that the entries in the LSR routing tables must be taken into account when assigning labels to packets and building switching tables.

The set of packets that would receive the same treatment by an LSR (i.e., packets that will be forwarded towards the same port of the same destination network) is called *Forwarding Equivalence Class* (FEC). LSRs bind FECs with label / port pairs. For example, all packets that must be delivered to the network 20.10.10.0/24 constitute an FEC that might be bound to the pair (4, 881). All packets directed to that network will be switched to the port 4 with label 881. The treatment that packets will receive on the next hop depends on the selection of the outgoing label. In our example, a packet switched to the port 4 with label 882 will probably never arrive to network 20.10.10.0/24. An LSR may need to request the right label at the next hop to ensure that the packets will receive the desired treatment and that they will be forwarded to the correct destination.

The most common FECs are defined by network address prefixes stored in the routing tables of LSRs. In the routing table, the network prefix determines the outgoing interface for the set of incoming packets matching this prefix. If we wish to emulate the behaviour of a traditional IP router, every network prefix must be bound with a label.

Within the MPLS domain, labels only have a local meaning, which is why the same label can be re-used by different LSRs. For the same packet, the value of the label can be different at every hop, but the path a packet follows in the network is totally determined by the label assigned by the ingressing LER. The sequence of labels [315, 422, 288] defines an LSP route, all packets following the LSP receive the same treatment in terms of bandwidth, delays, or priority enabling specific treatment to each traffic flow like voice, data or video. There are two LSP types (see Figure 10.19):

- *Hop-by-hop LSPs* are computed with routing protocols alone. MPLS networks with only hop-by-hop LSP route packets are like traditional IP networks but with enhanced forwarding performance provided by label switching.
- *Explicit LSPs* are computed by the network administrators to meet specific purposes, and configured either manually in the LSRs, or with the help of the management platform. The path followed by the packets forwarded across explicit LSPs may be different from the paths computed by routing protocols. They can be useful to improve network utilization or select custom paths for certain packets. The ability to provide explicit LSPs converts MPLS into a powerful traffic engineering tool.

An explicit LSP can be strict or loose, depending on how it is established:

- If all the hops that constitute the explicit LSP are specified one by one, the LSP is said to be strictly specified.
- If some but not all the hops that constitute the explicit LSP are specified but some others are left to the decision of the distributed routing algorithms, the LSP is said to be loosely specified.

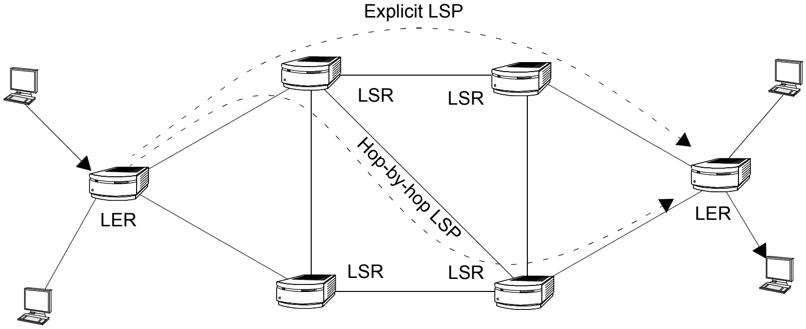


Figure 10.19 A hop-by-hop LSP and an explicit LSP between the same source and destination. The hop-by-hop LSP is computed by the routing protocols running in the LSRs. The explicit route is computed by an external *Network Management System* (NMS).

10.4.3 Label Distribution

The LSR needs to know which label to assign to outgoing packets to make sure they arrive to the correct destination. The obvious way to do this is to configure the switching tables manually in every LSRs. Of course, this approach is not the best possible if there are many LSPs dynamically established and released. To deal with this situation a label distribution protocol is needed.

A label distribution protocol enables an LSR to tell other LSRs the meaning of the labels it is using, as well as the destination of the packets that contain certain labels. In other words, by using a label distribution protocol the LSR can assign labels to FECs.

The RFC 3036 defines the *Label Distribution Protocol* (LDP) that was specifically designed for distributing labels. As MPLS technology evolved, this protocol showed its limitations:

- *It can only manage hop-by-hop LSPs*. It cannot establish explicit LSPs and therefore does not allow traffic engineering in the MPLS network.
- *It cannot reserve resources on a per-LSP basis*. This limits the QoS that can be obtained with LSPs established with LDP.

The basic LDP protocol is extended in RFC 3212 to support these and some other features. The result is known as the *Constraint-based Routed LDP* (CR-LDP). Another different approach is to extend an external protocol to work with MPLS. This is the idea behind the *ReSerVation Protocol with Traffic Engineering extension* (RSVP-TE) as defined in RFC 3209. The original purpose of the RSVP is to allocate and release resources along traditional IP routes, but it can be easily extended to work with LSPs. The traffic engineering extension allows this protocol to establish both strict and loose explicit LSPs.

10.4.3.1 The Label Distribution Protocol

The LDP enables LSRs to request and share MPLS labels. To do this it uses four different message types.

1. *Discovery messages* announce the presence of LSRs in the network. LSRs send “Hello” messages periodically, to announce their presence to other LSRs. These “Hello” messages are delivered to the 646 UDP port. They can be unicasted to a specific LSR or multicasted to all routers in the subnetwork.
2. *Session messages* establish, maintain and terminate sessions between LDP peers. To share label to FEC binding information, two LSRs need to establish an LDP session between them. Sessions are transported across the reliable TCP protocol and they directed to port 646.
3. *Advertisement messages* create, modify or delete label mappings for FECs. To exchange advertisement messages, the LSRs must first establish a session.
4. *Notification messages* are used to deliver advisory or error information.

The most important LDP messages are (see Figure 10.20):

- The *Label Request Message*, used by the LSR to request a label to bind with an FEC that is attached to the message. The FEC is commonly specified as a network prefix address.
- The *Label Mapping Message*, distributed by the LSR to inform a remote LSR on which label to use for a specific FEC.

The LSR can request a label for an FEC by using request messages, but it can also deliver labels to FEC bindings without explicit request from other LSRs. The former is an operation mode called *Downstream on Demand*, and the latter is known as *Downstream Unsolicited*. Both modes can be used simultaneously in the same network.

Regarding the behaviour of LSRs when they operate in the Downstream on Demand mode, receiving label request messages, there are two different options:

- *Independent label distribution control*: LSRs are allowed to reply to label requests with label mappings whenever they desire, for example immediately after the request arrives. This mode can be compared to the *Address Resolution Protocol* (ARP) used in LANs to request mappings between destination IP addresses and MAC addresses.

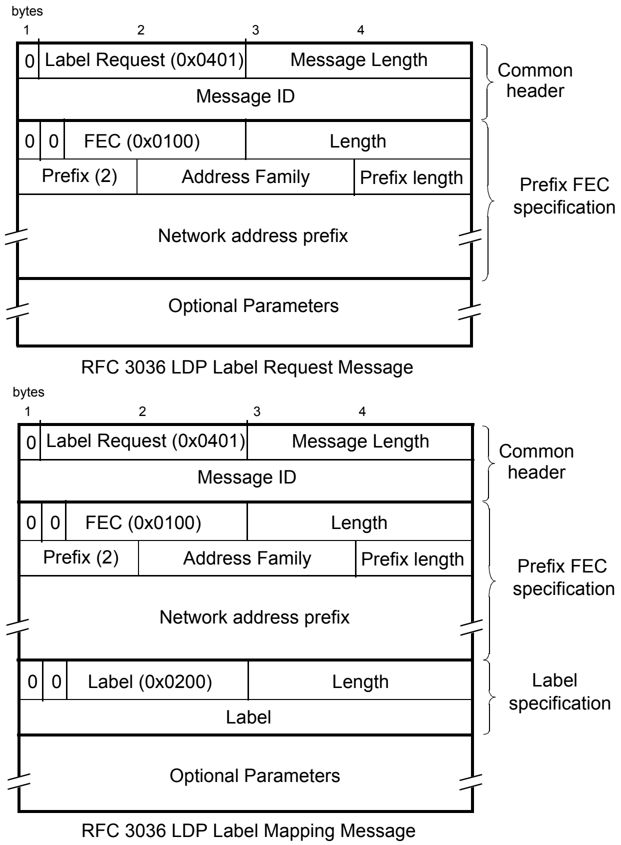
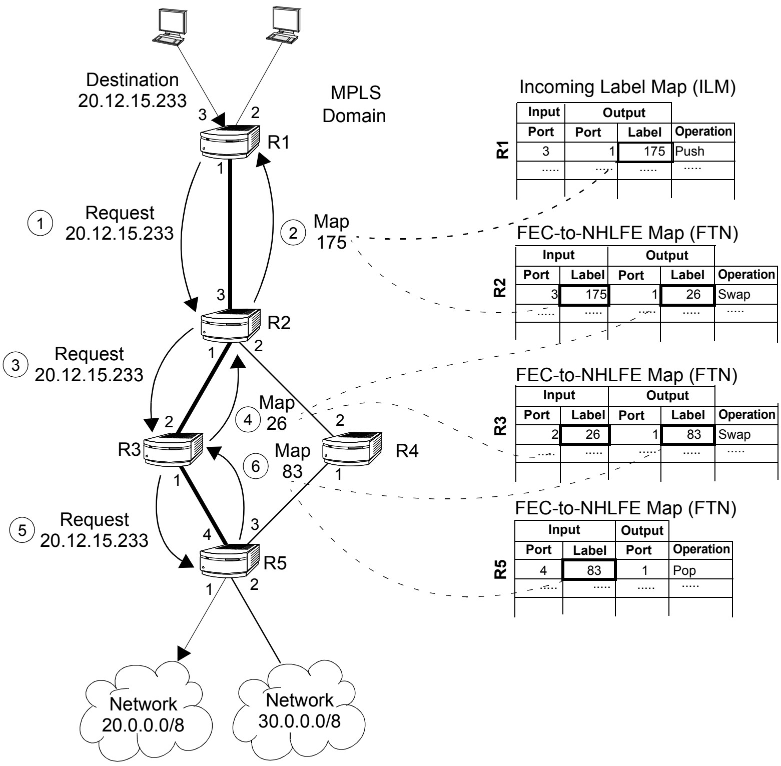


Figure 10.20 Two important LDP messages. The *Label Request Message* requests a label from a remote LSR for binding with an FEC that is attached to the message. The *Label Mapping Message* is used to inform a remote LSR on which label to use for a specific FEC.

- *Ordered label distribution control* (see Figure 10.21): LSRs are not allowed to reply to label requests until they know what to do with the packets belonging to the mapped FEC. In other words, LSRs cannot map an FEC with a label unless they have a label for the FEC, or if they are egress LERs themselves. When an LSR operates in this mode, it propagates the label requests downstream and waits for a reply before replying upstream.



Incoming Label Map (ILM)

	Input	Output		
	Port	Port	Label	Operation
R1	3	1	175	Push
	.....	.....	.....	.....

FEC-to-NHLFE Map (FTN)

	Input		Output		
	Port	Label	Port	Label	Operation
R2	3	175	1	26	Swap
	.....	.....	.....	.....	.....

FEC-to-NHLFE Map (FTN)

	Input		Output		
	Port	Label	Port	Label	Operation
R3	2	26	1	83	Swap
	.....	.....	.....	.....	.....

FEC-to-NHLFE Map (FTN)

	Input		Output	
	Port	Label	Port	Operation
R5	4	83	1	Pop
	.....	.....	.....	.....

Figure 10.21 LDP in Downstream-on-Demand and independent label distribution mode. LSRs in the LSP generate label requests. The replies they receive from LSRs upstream are used to fill their switching tables.

10.4.4 Martini Encapsulation

In the MPLS network, only the ingress and egress LERs are directly attached to the end-user equipment. This makes them suitable for establishing edge-to-edge sessions to enable communications between remote users. In this network model, the roles of LSRs and LERs would be:

- *LSRs* are in charge of guiding the frame through the MPLS network, using either IP routing protocols or paths that the network administrator has chosen by means of explicit LSPs.

- The *Ingress LER* is in charge of the same tasks as any other LSR, but it also establishes sessions with remote LERs to deliver traffic to the end-user equipment attached to them.
- The *Egress LER* acts as the peer of the ingress LER in the edge-to-edge session, but it does not need to guide the traffic through the MPLS network, because the traffic leaves the network in this node and it is not routed back to it.

There is an elegant way to implement the discussed model without any new overhead or signaling: by using label stacking. This model needs an encapsulation with a two-label stack known as the *Martini encapsulation* (see Figure 10.22):

- The *Tunnel label* is used to guide the frame through the MPLS network. This label is pushed by the ingress LER and popped by the egress LER, but it can also be popped by the penultimate hop in the path, because this LSR makes the last routing decision within the MPLS domain, thus making the Tunnel label unnecessary for the last hop (the egress LER).
- The *VC label* is used by the egress LER to identify client traffic and forward the frames to their destination. The way the traffic reaches end users is a decision taken by the ingress and egress nodes, and it does not involve the internal LSRs. The VC label is therefore pushed by the ingress LSR and popped by the egress LSR.

In the non-hierarchical one-label model, all the routers in the LSP participate in establishing an edge-to-edge session, and all are involved in routing decisions as well. A two-label model involves two types of LSPs. The tunnel LSP may have many hops, but the VC LSP has only two nodes, the ingress and egress LERs. VC LSPs can be interpreted as edge-to-edge sessions that are classified into groups and delivered across the MPLS network within Tunnel LSPs (see Figure 10.23). Tunnel LSPs are established and released independently of the VC LSPs. For example, Tunnel LSPs can be established or modified when new nodes are connected to the network, and VC LSPs could be set up when users wish to communicate between them.

The two-label model makes routing and session management independent of each other. It is not necessary to maintain status information about sessions in the internal LSRs. All these tasks are carried out by LERs. The signaling of the VC LSP is also different from that of the Tunnel LSP. While establishing a Tunnel LSP may require specific QoS or it may depend on administrative policies relying on traffic engineering, VC LSPs are much more simple. This is the reason why label distribution of Tunnel LSPs is carried out with the CR-LDP or the RSVP-TE protocols, but VC LSPs can be managed with the simple LDP.

Although the two-label approach is valid for any MPLS implementation, it has been defined to be used with pseudowires (see Paragraph 10.4.5).

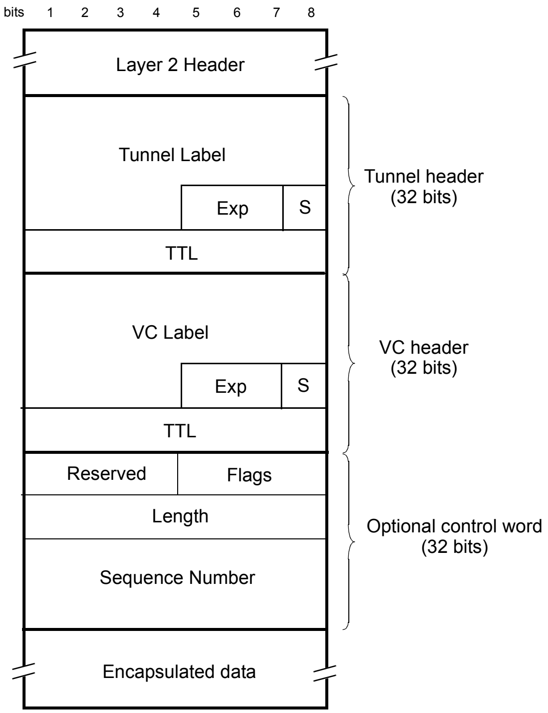


Figure 10.22 Two-label MPLS stack with a Tunnel label and a VC label. The control word may be required when carrying non-IP traffic.

10.4.5 Pseudowires

Pseudowires are entities that carry the essential elements of layer-2 frames or TDM circuits over a packet-switched network with the help of MPLS[2]. The standarization of pseudowires is driven by the demand of *Virtual Private Wire Services* (VPWS) that can transport Ethernet, FR, ATM, PPP, SDH, Fiber Channel and other technologies in a very flexible and scalable way. This fact moved the IETF to create the *Pseudowire Edge-to-Edge Emulation* (PWE3) working group that generates standards for encapsulations, signaling, architectures and applications of pseudowires.

The concept of pseudowire relies on a simple fact: within the MPLS network, only labels are used to forward the traffic, and any other field located in the payload that could be used for switching is ignored. This means that the data behind the

[2] Although it is possible to implement pseudowires without MPLS, it is used in all the important solutions due to its better performance when compared to other options.

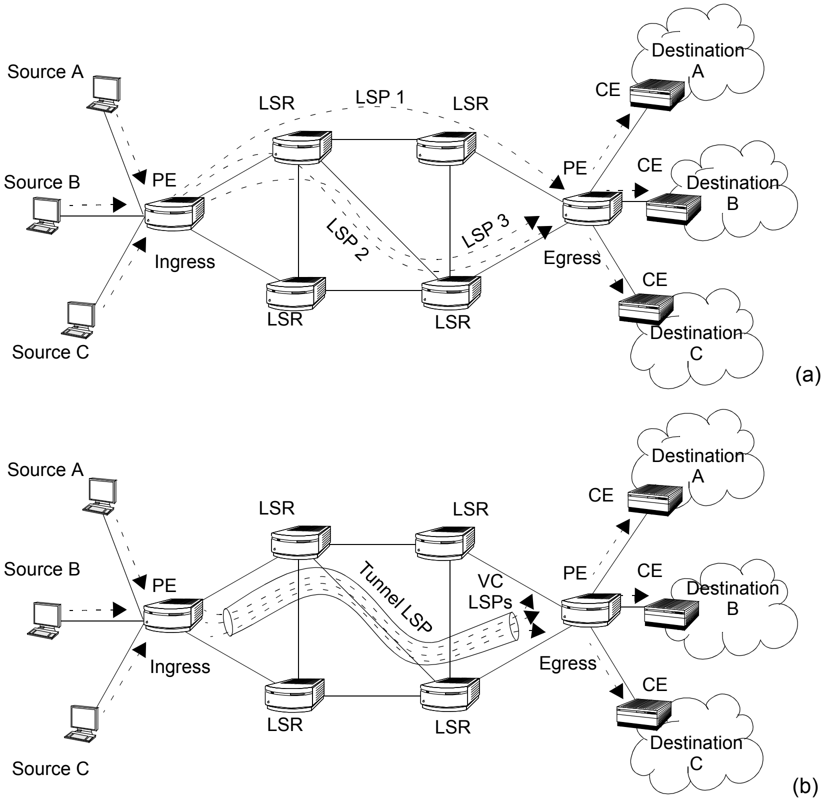
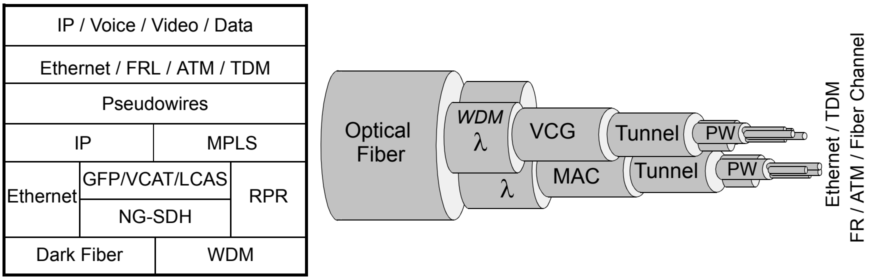


Figure 10.23 (a) One-label approach: the decision to establish routing and edge-to-edge sessions is shared between all the routers. (b) Two-label model: edge-to-edge sessions are tunneled, and internal LSRs are unaware of them.

MPLS header could be potentially anything, not limited to an IP datagram. The advanced QoS capabilities of MPLS, including resource management with the RSVP-TE or the CR-LDP protocols make it suitable for transporting traffic subject to tight delay and jitter constraints, including SDH and other technologies based on TDM frames (see Figure 10.24).S

It is worth noting that although in MPLS-based pseudowires IP datagrams are replaced by layer-2 or TDM data, IP routing is still an important part of the network. OSPF, IS-IS or other routing protocols are still necessary to find routes in the service provider network when they are not explicitly defined in the LSP setup process. This means that in the MPLS network carrying pseudowires, IP numbering must be maintained in the network interfaces, because IP routing protocols need IP addresses to work.



<table>
<tr><td colspan="4">IP / Voice / Video / Data</td></tr>
<tr><td colspan="4">Ethernet / FRL / ATM / TDM</td></tr>
<tr><td colspan="4">Pseudowires</td></tr>
<tr><td colspan="2">IP</td><td colspan="2">MPLS</td></tr>
<tr><td rowspan="2">Ethernet</td><td colspan="2">GFP/VCAT/LCAS</td><td rowspan="2">RPR</td></tr>
<tr><td colspan="2">NG-SDH</td></tr>
<tr><td colspan="2">Dark Fiber</td><td colspan="2">WDM</td></tr>
</table>

Figure 10.24 Pseudowires can encapsulate and transport ATM, FR, Ethernet, PPP, TDM or Fiber Channel, which is why these protocols do not need a dedicated network unifying the transport in one single network and interface.

Pseudowires are tunneled across the packet-switched network (see Figure 10.25). Any network capable of providing tunnels can be used as a transport infrastructure. By far, MPLS is the most common transport infrastructure for pseudowires, but pure IP networks can be used for the same purpose as well. The MPLS-based pseudowires use LSPs as tunnels, but IP networks need to use other tunnels, such as *Generic Routing Encapsulation* (GRE) or *Layer-2 Tunneling Protocol* (L2TP) tunnels.

Many pseudowires are allowed to be multiplexed in the same tunnel, and therefore it is necessary to identify them. An MPLS label can be used for this purpose, although the IP transport infrastructure uses different alternatives again. MPLS architectures need two labels for carrying pseudowires: the first to identify the tunnel and the second to identify the pseudowire. The tunnel / VC double labeling (see Paragraph 10.4.4) is applied to this case. Here, the VC label becomes the pseudowire identifier, and it is therefore known as the *PseudoWire* (PW) label.

In the traditional MPLS applications, FECs are specified by means of IP addresses or IP network prefixes. Once a label is bound with a network prefix, the network node automatically knows how to forward those packets that carry this label. However, this simple approach does not work with pseudowires, because they carry non-IP data. It is necessary to specify a new way to tell the pseudowire end points how to process the data carried by the pseudowire. This means that new ways of specifying FECs must be defined. Furthermore, each technology may need its own FEC specification. For example, forwarding Ethernet frames from or to pseudowires

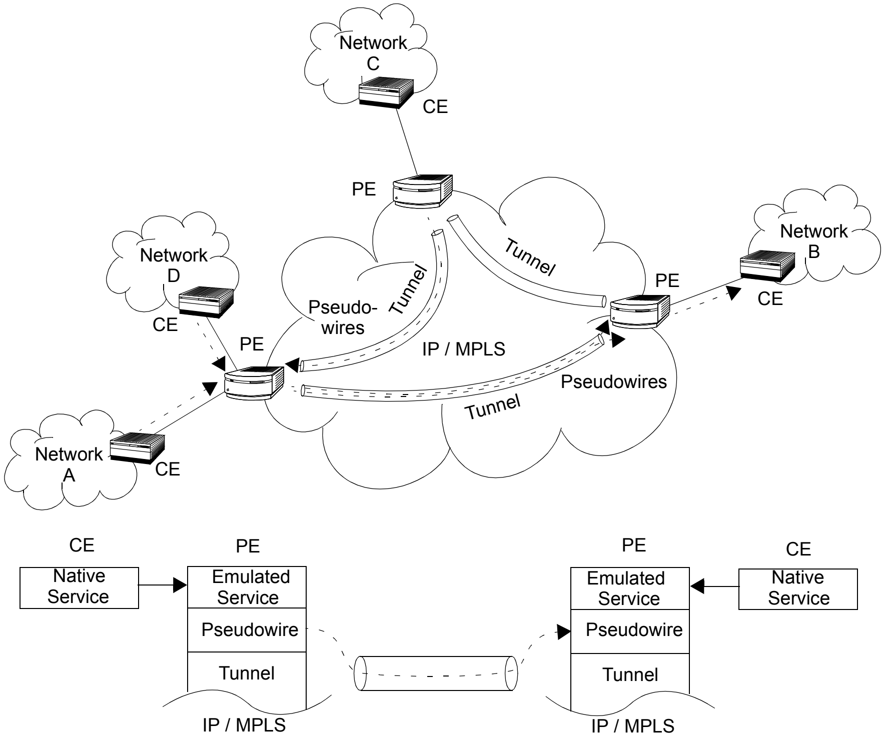


Figure 10.25 Emulation of connectivity services over pseudowires and tunneling across an IP / MPLS network.

depends on the physical port and the VLAN tag, but this is not necessarily true for ATM or SDH pseudowires. This problem is addressed by extending the LDP protocol to work with pseudowires (see Figure 10.26).

The existing definitions are generalistic and have different interpretations for different types of pseudowire. This is the reason why the new FEC specifications include a 16-bit field for choosing the service emulated over the packet-switched network (see Table 10.3).

Sometimes, it must be ensured that packets are received in the correct order. Other times it is necessary to pad small packets with extra bits, or add technology-specific control bits. To deal with these issues, an extra 32-bit word is inserted between the PW label and the encapsulated data (see Figure 10.22). The presence of this control word is sometimes required, other times optional, and occasionally not required at all, depending on the type of pseudowire used. The presence of the control word is signaled in the LDP protocol when the pseudowire is established.

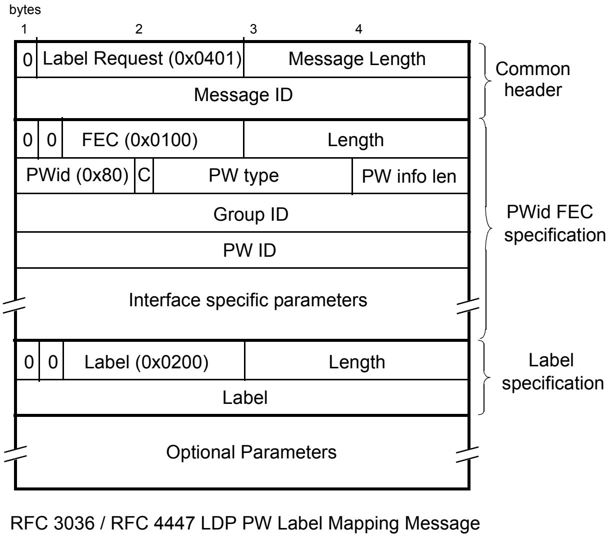


Figure 10.26 The LDP Label-Mapping message as used to map a pseudowire to an MPLS label. The label-to-pseudowire binding is done by using the PWid FEC element specified in RFC 4447.

Table 10.3
The existing types of pseudowire

PW type	*Description*
0x0001	Frame Relay DLCI (Martini mode)
0x0002	ATM AAL5 SDU VCC transport
0x0003	ATM transparent cell transport
0x0004	Ethernet tagged mode
0x0005	Ethernet
0x0006	HDLC
0x0007	PPP
0x0008	SONET / SDH Circuit Emulation Service over MPLS
0x0009	ATM n-to-one VCC cell transport
0x000a	ATM n-to-one VPC cell transport
0x000b	IP Layer2 transport
0x000c	ATM one-to-one VCC cell mode
0x000d	ATM one-to-one VPC cell mode

Table 10.3
The existing types of pseudowire

PW type	*Description*
0x000e	ATM AAL5 PDU VCC transport
0x000f	Frame Relay port mode
0x0010	SONET / SDH circuit emulation over packet
0x0011	Structure-agnostic E1 over packet
0x0012	Structure-agnostic T1 (DS1) over packet
0x0013	Structure-agnostic E3 over packet
0x0014	Structure-agnostic T3 (DS3) over packet
0x0015	CESoPSN basic mode
0x0016	TDMoIP AAL1 mode
0x0017	CESoPSN TCM with CAS
0x0018	TDMoIP AAL2 mode
0x0019	Frame Relay DLCI

10.4.6 Ethernet Pseudowires

The aim of Ethernet pseudowires is to enable transport of Ethernet frames across a packet-switched network and emulate the essential attributes of Ethernet LANs, such as MAC frame bridging or VLAN filtering across that network.

Standardization of pseudowires enables IP / MPLS networks to transport Ethernet efficiently. The Ethernet pseudowire is perhaps the most important type of pseudowire, because it can be used by network operators to fix some of the scalability, resilience, security and QoS problems of provider Ethernet bridges, thus making it possible to offer a wide range of carrier grade, point-to-point and multipoint-to-multipoint Ethernet services, including EPL, EVPL, EPLAN and EVPLAN.

PE routers with Ethernet pseudowires can be understood as network elements with both physical and virtual ports. The physical ports are the attachment circuits where CEs are connected through standard Ethernet interfaces. The virtual ports are Ethernet pseudowires. Frames are forwarded to physical or virtual ports, depending on their incoming port and VLAN tags. These network elements may also include flooding and learning features to bridge frames to and from physical ports and Ethernet pseudowires, thus making it possible to offer emulated multipoint-to-multipoint LAN services. Many of these PE routers are also able to shape and police Ethernet traffic to limit traffic ingressing in the service provider network.

When a new PE router is connected to the network, it must create tunnels to reach remote PE routers. The remote router addresses may be provided by the network administrators but many PE routers have autodiscovery features. Once the tun-

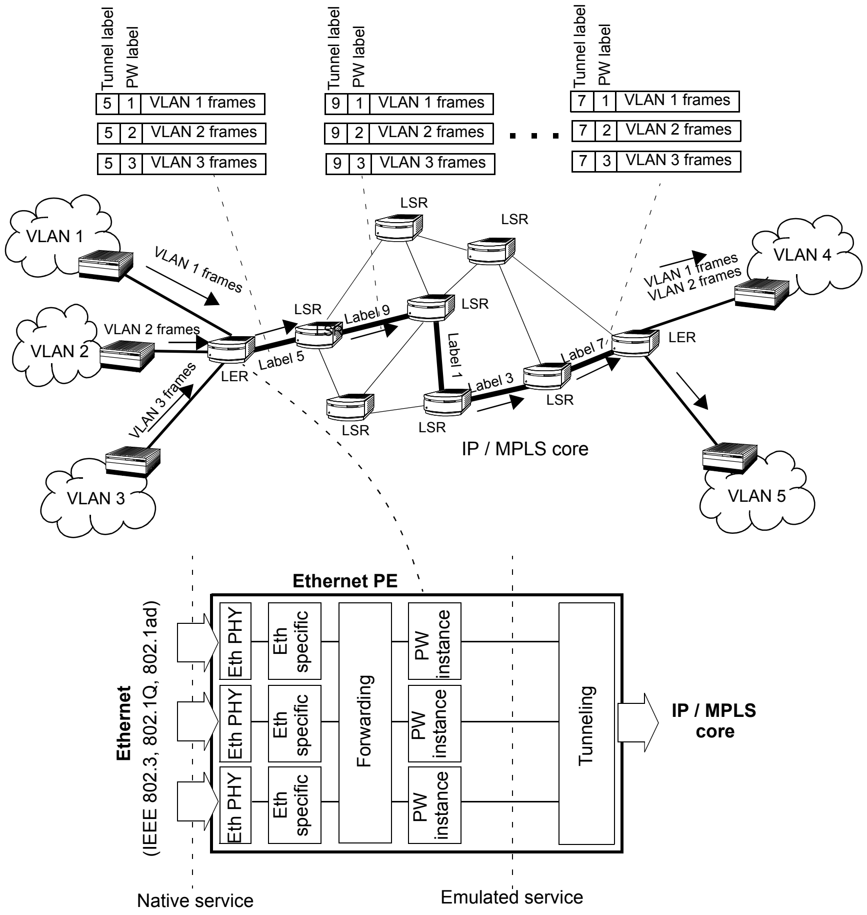


Figure 10.27 Operation of Ethernet pseudowires. The PE router becomes an Ethernet bridge with physical and virtual ports. Physical ports are connected to CEs with standard Ethernet interfaces. Virtual ports are Ethernet pseudowires tunneled across the IP / MPLS core.

nels are established, it is possible to start the pseudowire setup with the help of LDP signaling. LDP mapping signals tell the remote PE routers to which physical port and to which VLANs frames with specified PW labels (see Figure 10.27) will be switched.

The physical attachment circuits of the PE router are standard Ethernet interfaces. Some of them may be trunk links with VLAN-tagged MAC frames, or even double VLAN-tagged Q-in-Q frames. Regarding how VLAN tags are processed, the PE routers have two operation modes:

- *Tagged mode*: The MAC frames contain at least one service-delimiting VLAN tag. Frames with different VLAN IDs may belong to different customers, or if they belong to the same customer, they may require different treatment in the service provider network. MAC frames with service-delimiting VLAN tags may be forwarded to different pseudowires or mapped to different Exp values for custom QoS treatment.
- *Raw mode*: The MAC frames may contain VLAN tags, but they are not service-delimiting. This means that any VLAN tag is part of the customer VLAN structure and must be transparently passed through the network without processing.

10.4.6.1 Virtual Private LAN Service

The *Virtual Private LAN Service* (VPLS) is a multipoint-to-multipoint service that emulates a bridged LAN across the IP / MPLS core.

VPLS is an important example of a layer-2 *Virtual Private Network* (VPN) service. Unlike more traditional layer-3 VPNs, based on network layer encapsulations and routing, layer-2 VPNs are based on bridging to connect two or more remote locations as if they were connected to the same LAN. Layer-2 VPNs are simple and well suited to business subscribers demanding Ethernet connectivity. VPLS also constitutes a key technology for metropolitan networks. This technology is currently available for network operators who want to provide broadband triple play services to a large number of residential customers.

When running VPLS, the service provider network behaves like a huge Ethernet switch that forwards MAC frames where necessary, learns new MAC addresses dynamically, and performs flooding of MAC frames with unknown destination. In this architecture, PE routers behave like Ethernet bridges that can forward frames both to physical ports and pseudowires.

As with physical wires, bridging loops may also occur in pseudowires. If fact, it is likely that this occurs if the pseudowire topology is not closely controlled, because pseudowires are no more than automatically established LDP sessions. A bridged network cannot work with loops. Fortunately, the STP or any of its variants can be used with pseudowires, as is done with physical wires to avoid them. However, there is another approach recommended by the standards. The most dangerous situation occurs when a PE router relays MAC frames from a pseudowire to a second pseudowire. To avoid pseudowire-to-pseudowire relaying, a direct pseudowire connection must be enabled between each PE router in the network. This implies a full-mesh pseudowire topology (see Figure 10.28). The full-mesh topology is com-

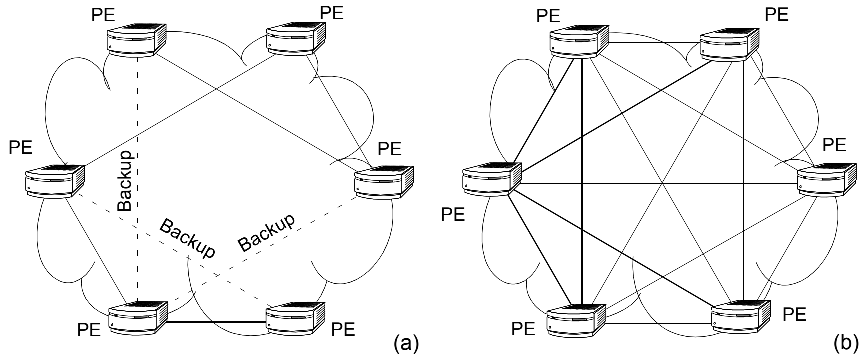


Figure 10.28 Pseudowire topologies in VPLS: (a) Partial mesh with STP. Some of the pseudowires are disabled to avoid loops. (b) Full mesh of pseudowires. The split-horizon rule is applied to avoid bridging loops.

pleted with the *split-horizon rule*: It is forbidden to relay a MAC frame from a pseudowire to another one in the same VLPS mesh. Relaying would any way be unnecessary because there is a direct connection with every possible destination.

To understand how VPLS works we can think of two end users, S and D, who want to communicate to each other (see Figure 10.29). User S wants to send a MAC frame to user D across a shared network running VPLS.

1. S sends the MAC frame towards D. LAN A is unable to find a local connection to D and finally the frame reaches bridge CE 1 that connects LAN A to a service provider network.
2. Bridge CE 1 forwards S's frame to PE 1 placed at the edge of a VPLS mesh. If PE 1 has not previously learnt S's MAC address, it binds it with the physical port where the frame came from.
3. The PE 1 bridge has not previously learnt the destination address of the MAC frame (D's MAC address), and therefore it floods the frame to all its physical attachment circuits. S's frame reaches LAN B, but D is not connected to it.
4. PE 1 not only performs flooding on its physical ports, but also on the pseudowires. S's frame is thus forwarded to all other PEs in the network by means of direct pseudowire connections across the VPLS mesh.
5. S's frame reaches PE 2 attached to pseudowire PW12. If PE 2 has not previously learnt the received source MAC address, it binds it with pseudowire PW12. In this case, PE 2 does not know where D is, so it flows the MAC frame to all the physical ports and arrives to LAN C, however D is not connected to that LAN. Following the split-horizon rule, the frame is not flooded to other pseudowires.
6. S's frame reaches PE 4. It learns S's MAC address if it is unaware of it. After

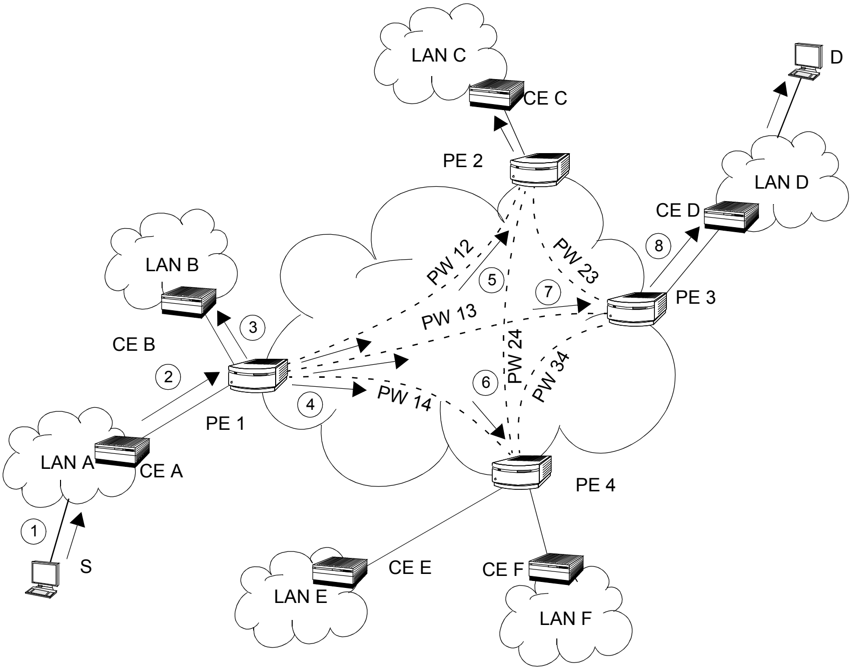


Figure 10.29 Flooding and learning in VPLS serves emulate a LAN broadcast domain

learning, S's address is bound to pseudowire PW14. In this case PE 4 has previously bounded D's address to pseudowire PW34, and therefore it does not forward S's frame to LAN E or LAN F. The frame is not forwarded to pseudowire PW 4 either, because of the split-horizon rule.

7. S's frame reaches PE 3. This router performs the same learning actions as PE 2 and PE 4 if needed, and binds S's MAC address to pseudowire PW13. In this case, PE 3 has previously learnt that D can be reached by one of its physical ports, and therefore it forwards S's frame to it.
8. S's frame reaches CE D that forwards this frame to its final destination.

The previous example deals with a single broadcast domain that appears as a single distributed LAN. But this may not be acceptable when providing services to many customers. Every customer will normally require its own broadcast domain. The natural way to solve this is by means of VLANs. Every subscriber is assigned a service-delimiting VLAN ID. Every VLAN is then mapped to a VPLS instance (i.e., a broadcast domain) with its own pseudowire mesh and learning tables. The link between CE and PE routers is multiplexed, and customers are identified by VLAN tags. This deployment is useful for offering EVPLAN services as defined by the MEF.

But VLAN tags are not always meaningful for the service provider network. All VLAN tags can be mapped to a single VPLS instance and therefore all of them are part of the same broadcast domain within the service provider network. In this case VLAN-tagged frames are filtered by the subscriber network, but they are leaved unchanged in the service provider network. Different customers can still be assigned to different broadcast domains, but not on a per-VLAN-ID basis. Mapping customers to VPLS instances on a per-physical-port basis is the solution in this case. This second deployment option is compatible with the EPLAN connectivity service definition given by the MEF.

VPLS has demonstrated to be flexible, reliable and efficient, but it still lacks scalability due excessive packet replication and excessive LDP signaling. The origin of the problem is on the full meshed pseudowire topology. The total number of pseudowires needed for a network with n PE routers is $n(n-1)/2$. This limits the maximum number of PE routers to about 60 units with current technology.

Hierarchical VPLS (HVPLS) is an attempt to solve this problem by replacing the full meshed topology with a more scalable one. To do this it uses a new type of network element, the *Multi-Tenant Unit* (MTU). In HVPLS, the pseudowire topology is extended from the PE to the MTU. The MTU now performs some of the functions of the PE, such as interacting with the CE and bridging. The main function of the PE is still frame forwarding based on VLAN tags or labels. In some HVPLS architectures, the PE does not implement bridging. The result is a two-tier architecture with a full mesh of pseudowires in the core and non redundant point-to-point links between the PE and the MTU (see Figure 10.30). A full mesh between the MTUs is not required, and this reduces the number of pseudowires. The core network still needs the full mesh, but now the number of PEs can be reduced, because some of their functions have been moved to the access network.

The MTUs behave like normal bridges. They have one (and only one) active pseudowire connection with the PE per VPLS instance. Flooding, as well as MAC address learning and aging is performed in the pseudowire as if it were a physical wire. The PE operates the same way in an HVPLS as in a flat VPLS, but the PE-MTU pseudowire connection is considered as a physical wire. This means that the split-horizon rule does not apply to this interface.

In practical architectures, the MTUs are not always MPLS routers. Implementations based on IEEE 802.1ad service provider bridges are valid as well. These bridges make use of Q-in-Q encapsulation with two stacked VLAN tags. One of

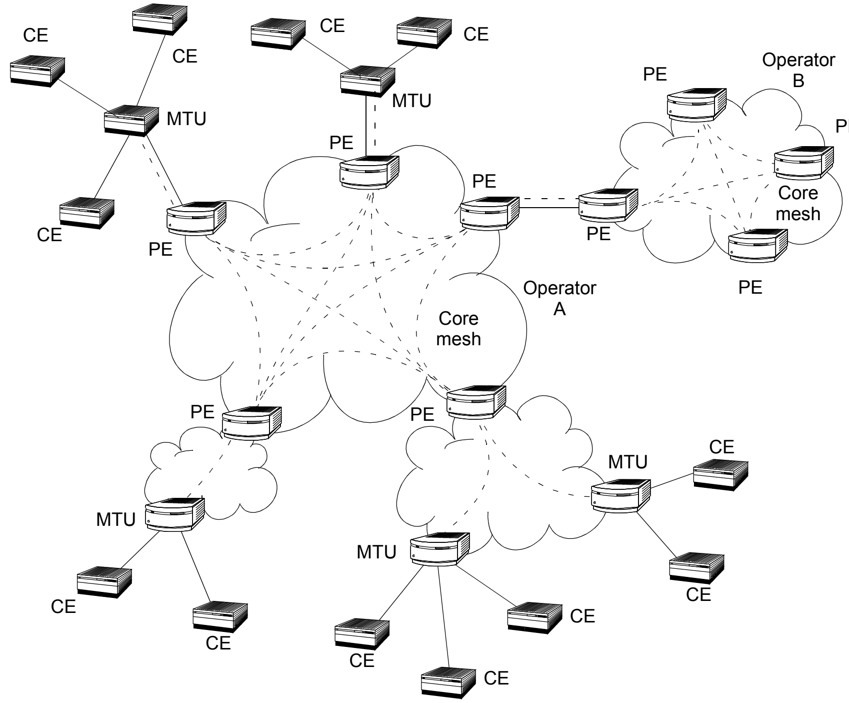


Figure 10.30 In HVPLS, the full mesh of pseudowires is replaced by a two-tier topology with full mesh only in the core and non-redundant point-to-point links in the access.

these tags is the service delimiting P-VLAN tag added by the MTU. The P-VLAN designates the customer, and it is used by the PE for mapping the frames to the correct VPLS instance.

HVPLS can be used to extend the simple VPLS to a multioperator environment. In this case, the PE-MTU non-redundant links are replaced by PE-PE links where each PE in the link belongs to a different operator.

The main drawback of the HVPLS architecture is the need for non-redundant MTU-PE pseudowires. A more fault tolerant approach would cause bridging loops. One solution is a multi-homed architecture with only one simultaneous MTU-PE pseudowire active. The STP can help in managing active and backup pseudowires in the multi-homed solution.

10.4.7 PWE3 and NG-SDH

The VPLS is the enabling technology for QoS and protection, allowing some operators to build SDH-free backhaul networks, especially if their traffic is only IP. However, PWE3 is being implemented in NG-SDH networks as well as in those cases where there are high QoS, TDM or OAM requirements.

The use of pseudowires over NG-SDH results in an efficient network that combines the MPLS features with the carrier-class approach of SDH. Edge MSSP nodes push labels to the entering traffic, or they pop them for the egressing traffic. Intermediate LER nodes check the labels at the input VCGs, swap the labels, and then switch the packets to the correct output VCG.

10.4.8 Advantages of the MPLS

IP routers operate by matching the IP address with its internal tables to calculate the next hop before the packets reaches its destination. In IP networks, all packets with same destination are treated in a very similar way. The MPLS adopted virtual circuit connections that help to set up paths that can provide QoS.

MPLS was created to optimize packet forwarding in IP networks to meet the required QoS, as routers had become a bottleneck. The original aim of the MPLS was to reduce processing time to make routing more efficient. Although the original intention is now irrelevant, the MPLS still has many important advantages that make it a key piece in the puzzle (see Figure 10.31):

1. *MPLS is protocol-agnostic*

- SDH, NG-SDH, Ethernet, Dark Fiber, WDM can be used as transport layer
- Most of the protocols can be encapsulated like Ethernet, FR, ATM, PPP, etc.
- Interoperatibility between different vendors is easier and more direct

2. *MPLS is a traffic engineering tool*

- MPLS paths can be pre-established to route traffic and avoid bottlenecks, or to give priorities
- Data flows can be separated to provide privacy for each customer's traffic
- A network with MPLS can provide different types of QoS
- MPLS can set up protection mechanisms.

10.5 MIGRATION

For most of today's communications services, telecom operators, carriers, and providers have SDH networks supplied with TDM circuits. The emerging demand for Ethernet-based services is pushing the migration to those services, as well as the re-

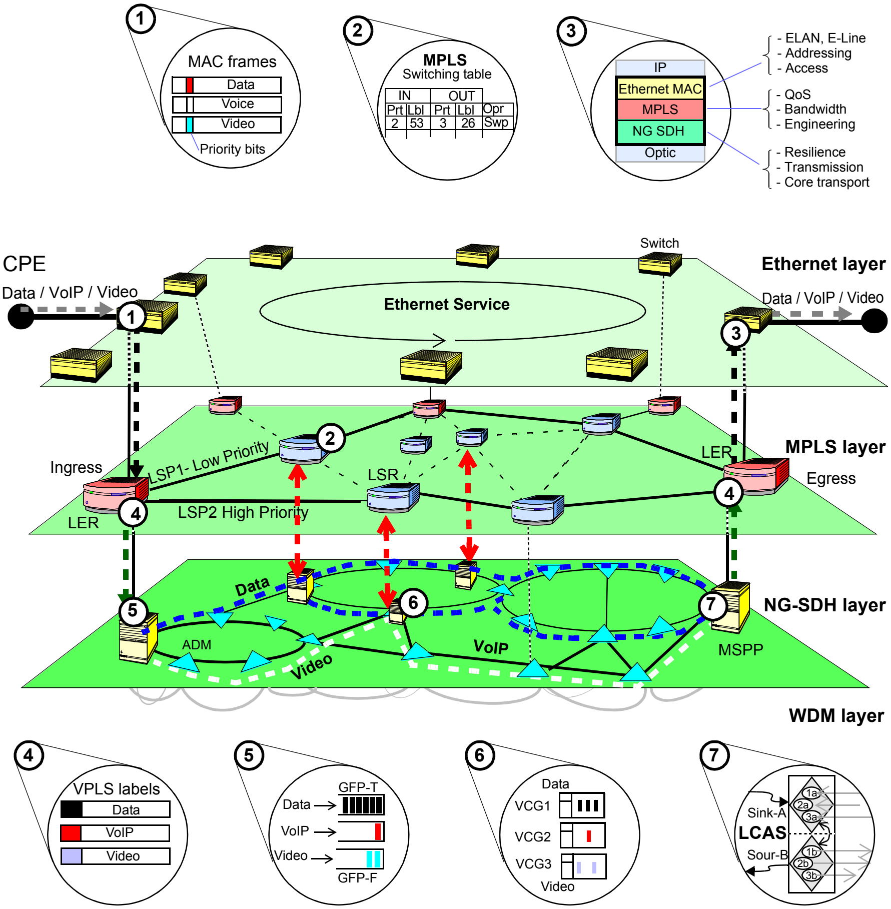


Figure 10.31 Ethernet customer traffic is sent to the service provider's MPLS-domain LER. The label attached to each packet defines an MPLS path with a certain QoS. Packets are then passed to one of the available VCGs according to the priority of the label. Immediately NG-SDH transmits the packet to the next LSR that checks the label again, and forwards the packet accordingly. When the packet finally reaches the LER, the encapsulation with the label is removed and the MAC frame is delivered to its destination.

use of the existing infrastructures. The path and the tempo of such migration depends on many factors, so each case should be studied separately. However, any migration route analysis should consider the following:

- Market demand, to identify which connectivity services are required, at which price and what is the cost for supplying them (CAPEX).
- Competition analysis, to define a realistic business case.

- Existing network architecture, including topologies and bit rates.
- Optical resources, spare fiber or wavelength may make it more attractive to use an overlapped network rather than a network integrated into SDH.
- Existing and new traffic.

10.5.1 Migrating the Architecture

Most of today's service providers rely on PDH/SDH networks (see Figure 10.32).

1. The starting architecture is an SDH network providing only TDM services. Ethernet services end where the routers are. Some are also mapped on TDM.
2. MSSP is used to allow native Ethernet interfaces, and submultiplexing is possible to increase the efficiency of packet services.
3. Ethernet switches are installed to increase flexibility and multi-point connections simplifying the NMS operation. It also enables layer-2 VPN.

10.5.2 Legacy Services

Legacy services are a combination of circuit and packet technologies:

- LAN: Ethernet is the standard
- Access: several technologies are used including DSL, E1, *n* x 64
- Voice: supplied in TDM by means of POTS and ISDN
- Data: transported by FRL and ATM
- Core: SDH is the most common transmission technology
- Core switching: ATM is often used for data from DSL access and Internet
- Optical: WDM is used in for high demand

10.5.3 Introduction to NG-SDH + Ethernet

The NG-SDH nodes (MSPP, MSTP) offer a combination of new data-packet and legacy interfaces, and include new features to efficiently support any type of traffic. This means that the Ethernet / IP tandem can obtain the most remarkable features of SDH, including resiliency, reliability, scalability, built-in protection, management and re-routing.

This architecture makes it possible to maintain all legacy networks and applications, plus the new packet-based networks. For Ethernet services only the MEF *Ethernet Private Line* (EPL) is possible, as the service can only be provided between two edge points.

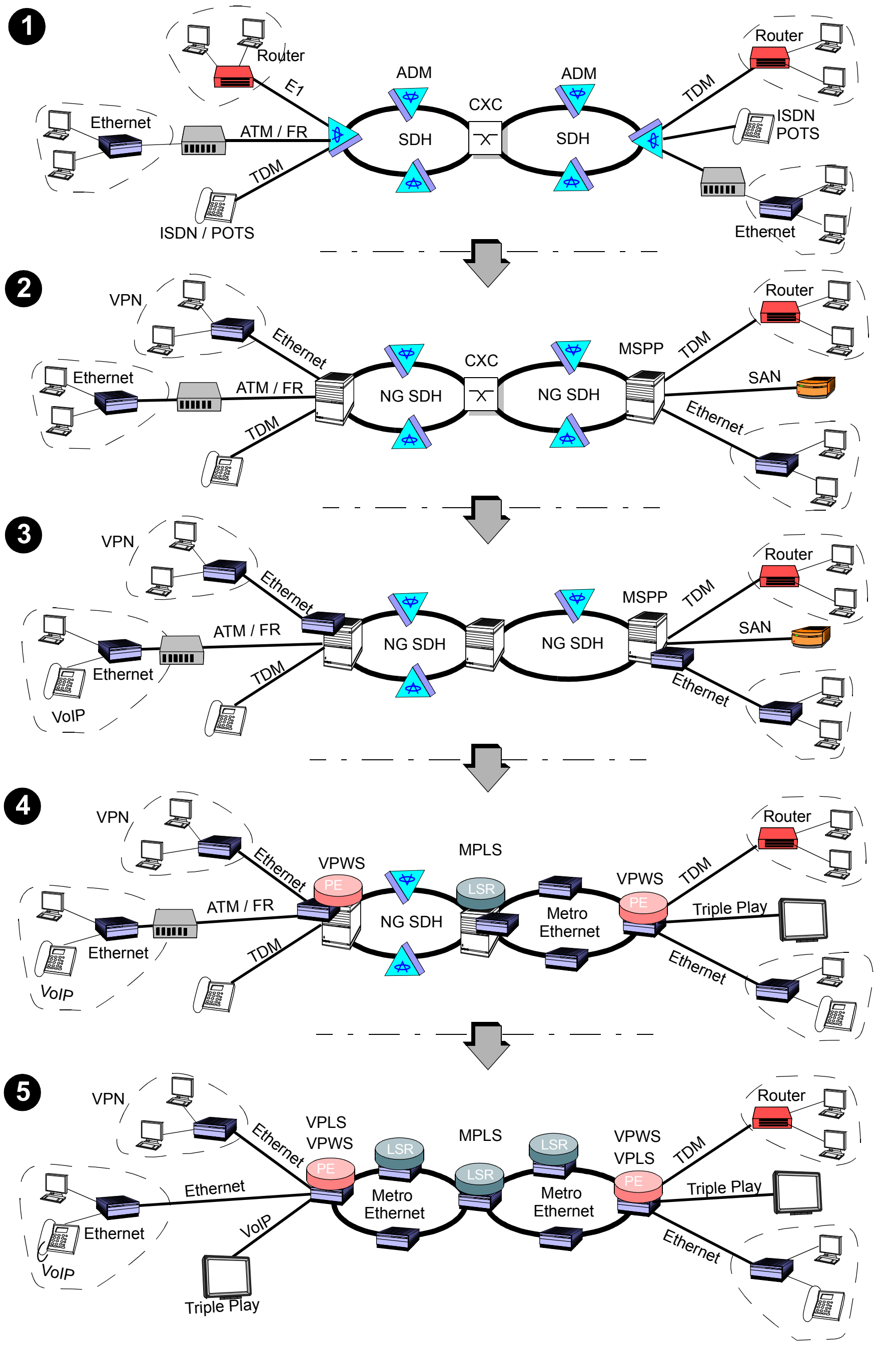


Figure 10.32 Migration strategies from a circuit oriented (TDM) to a packet (STDM) network.

10.5.4 NG-SDH + Ethernet Virtual Services

Adding Ethernet switches to the NG-SDH makes multipoint connectivity possible. This facilitates the support of not only EPL but also *Ethernet LAN* (ELAN), as well as virtual services (EPVL and EVLAN).

However, in the best of the cases, this architecture will be similar to a large Ethernet LAN with all the advantages and disadvantages of layer-2 networks, including scalability issues and the lack of a proper QoS-enabled service.

10.5.5 NG-SDH + MPLS + PWE3 + Ethernet

Including MPLS as unifying technology with PWE3 makes it possible to:

- Map ATM, FRL, PPP and TDM onto MPLS
- Provision QoS to support Triple Play services over Ethernet
- Increase scalability after fixing the MAC address explosion issue
- Use MPLS-based protection architectures

10.5.6 Service Interworking

The new architectures can provide a smooth migration from full TDM-oriented solutions to full packet-oriented solutions. But that is a long journey, and it will depend on many circumstances, including legacy installations, type of customers, applications, and which is the model that has been adopted for the new architecture.

When the data network is based on TDM, connection-oriented technologies like FRL, ATM and leased lines, the use of layer-2 MPLS may mean a smooth migration to packet services. Large corporations can roll out Ethernet on selected premises that may interact with the existing TDM networks:

- *Ethernet- FRL*: RFC 2427 Multi-protocol Interconnect over Frame Relay
- *Ethernet-ATM*: RFC 2684 Multi-protocol Encapsulation over ATM Adaptation Layer 5

10.5.7 Ethernet + MPLS – urbi et orbe?

In the short or middle term, will Ethernet be the only technology providing any type of service? We do not know, but consider this:

- Ethernet has always been a winner
- Historically, telecom services have had a long life

Statistical multiplexing everywhere? May be. We do not know yet, but it is true that there is a constant tendency to substitute TDM with packet networking at layer 3, layer 2 and layer 1 as well. Everywhere except at the optical layer.

Nevertheless, we should not underestimate the persistence of telecom technologies. Changes never take place suddenly, as we have seen during the past 30 years. Evolution and migration are always possible, whenever the installed base is not forced to perform a quick change, but a smooth one. This means that the integration and convergence of existing services must be guaranteed during migration. If Ethernet can do this, it is possible that in 10 years time most of the metro and core networks will use the technology that LANs made universal.

Chapter 11

Next Generation SDH/SONET

11.1 STREAMING FORCES

The financial and technological cycle of the telecoms industry is forcing manufacturers, carriers, operators and standards organizations to move towards a new network that reduces costs while offering new services.

The new services, mostly relying on data packet technology, offer easy implementation and access to applications based on Internet, mobile, multimedia, DVB, SAN, Ethernet or VPN. The architectures are increasingly requiring long-haul transport. However, packet technologies, like Ethernet, are in an early stage of development for efficient optical transport. SDH/SONET has such a massive installed base that carriers cannot simply get rid off it, for many reasons:

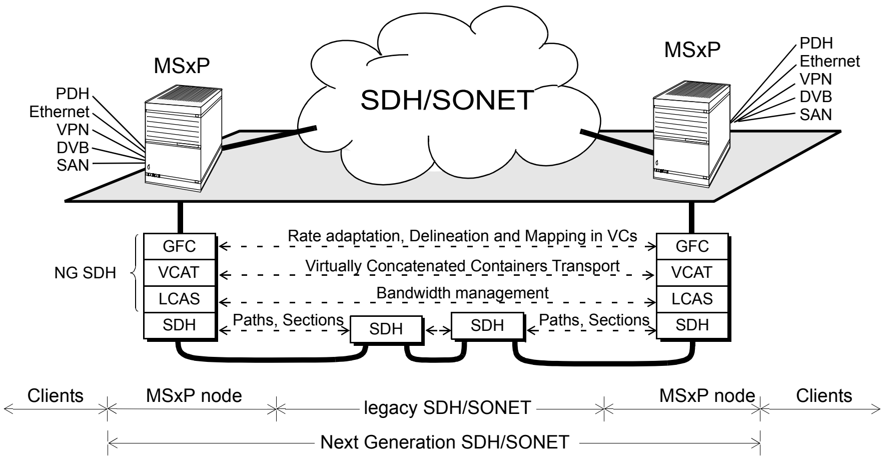


Figure 11.1 Next Generation SDH/SONET enables operators to provide more data transport services while increasing the efficiency of installed SDH/SONET base, by adding just the new MSxP edge nodes. This means that it will not be necessary to install an overlap network or migrating all the nodes or fiber optics. This reduces the cost per bit delivered, and will attract new customers while keeping legacy services.

- Most have their transport infrastructure entirely based on SDH/SONET.
- There is a lot of experience in managing SDH/SONET.
- No other technology has this maturity grade at the optical physical layer.

Luckily, SDH/SONET has also evolved to more efficiently adapt statistical multiplexing traffic based on data packets.

11.2 Legacy and Next Generation SDH

Telecom service providers are ready to include Ethernet/IP on their portfolio of solutions provided to the enterprise. This does not mean exactly that core network will soon be part of the past. No, it really means that a new generation of SDH/SONET nodes is offering a comprehensive combination of data-packet and TDM interfaces, optical physical layers (often based on DWDM), and a number of new functionalities to support efficiently any type of traffic (see Section 11.3.1). It also means that the Ethernet/IP tandem can obtain the most remarkable features of SDH/SONET including resiliency, reliability, scalability, built-in protection, management and rerouting.

11.2.1 Evolution of the Transmission Network

Most of the carriers and operators have been using SDH/SONET for several decades, mainly to transport voice and circuit-oriented data protocols. Since then, one of the challenges has been to efficiently transport statistically multiplexed, packet-oriented data services. Despite a number of architectures developed to do this, none of them were widely accepted by the market. Sometimes it was because of the cost, other times because of the complexity, and sometimes because of the poor efficiency.

Now, with the adoption of NG SDH/SONET, emerges a new opportunity driven by two factors: first, its simple encapsulation method, capable of accommodating any data packet protocols, and secondly, its demonstrated bandwidth efficiency. This means that a new adaptation protocol has been developed, as well as a new mapping mechanism for controlling bandwidth, while keeping the reliability of the legacy SDH/SONET transport and its centralized management.

11.3 The Next Generation Challenge

Compared to the legacy SDH/SONET, the Next Generation offers a much more flexible architecture which has turned out to be friendly with data packet networks,

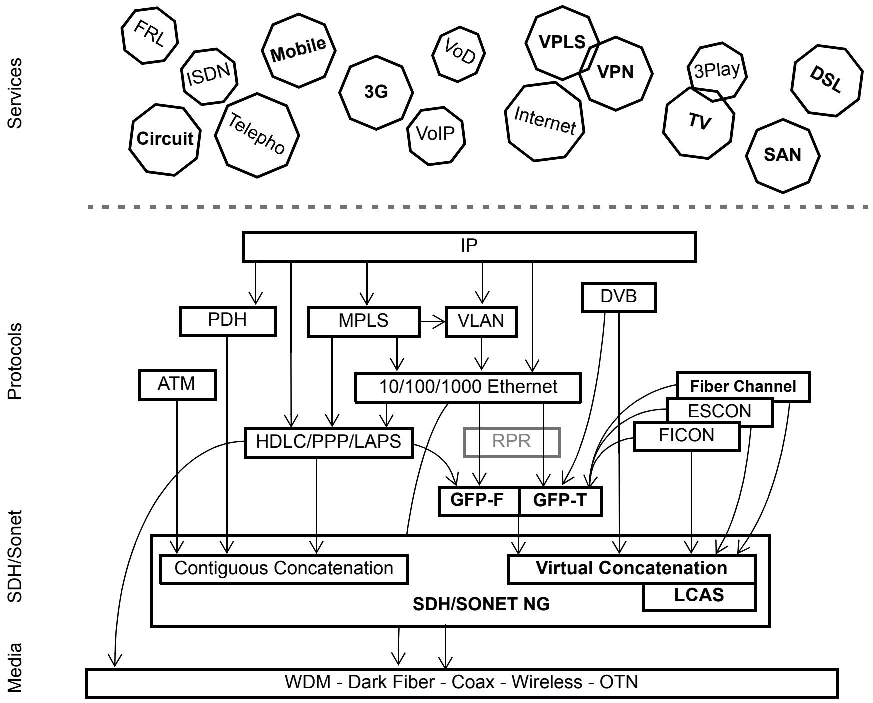


Figure 11.2 Versatile, flexible and efficient SDH Next Generation

and not only with circuit-oriented networks. Three new protocols have made this kind of evolution possible:

1. *Generic Framing Procedure* (GFP) is a robust and standardized encapsulation procedure for the transport of packetised data on SDH/SONET and OTN as well. In principal, GFP performs bit rate adaptation managing features, such as priorities, discard eligibility, transport channel selection and submultiplexing (see Figure 11.6). GFP replaces legacy mappings, many of them of proprietary nature.
2. *Virtual Concatenation* (VCAT) can create channels or pipes of granular bandwidth sizes rather than the exponential bandwidth provision of the contiguous concatenation. Therefore VCAT is more flexible and efficient.
3. *Link Capacity Adjustment Scheme* (LCAS) can modify the allocated VCAT bandwidth channel dynamically by adding/removing members of those channels that are already in use. LCAS overtakes the use of predefined bandwidth allocation, which does not match the variable bit rate patterns and the bursty nature of most data networks.

11.3.1 The New Network Elements

Under the generic name of *Multiservice Platforms* (MSxP) we find a new set of nodes that, combined with the legacy REG, ADM and DXC network elements, configure the topologies of the Next Generation SDH/SONET.

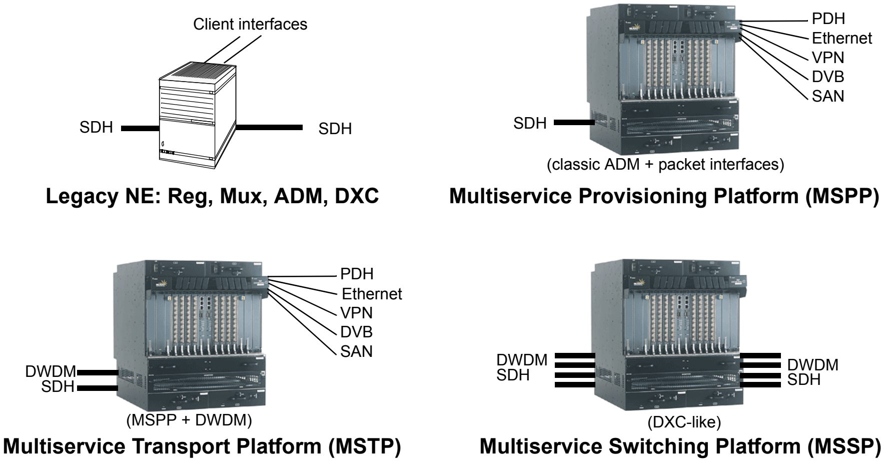

Client interfaces
SDH
SDH
Legacy NE: Reg, Mux, ADM, DXC
PDH
Ethernet
VPN
DVB
SAN
SDH
(classic ADM + packet interfaces)
Multiservice Provisioning Platform (MSPP)
PDH
Ethernet
VPN
DVB
SAN
DWDM
SDH
(MSPP + DWDM)
Multiservice Transport Platform (MSTP)
DWDM
SDH
DWDM
SDH
(DXC-like)
Multiservice Switching Platform (MSSP)

Figure 11.3 NG SDH/SONET network elements.

The MSxP nodes have a selection of data, circuit and optical interfaces, and they support the new GPF, VCAT and LCAS protocols. They can be classified as three types: MSPP, MSTP, and MSSP (see Figure 11.3).

11.3.1.1 Multiservice Provisioning Platform

A *Multiservice Provisioning Platform* (MSPP) is the result of the evolution of legacy ADM with TDM interfaces and optical interfaces, to a type of access node that includes a set of:

- legacy TDM, or circuit, interfaces
- data interfaces, such as Ethernet, GigE, Fiber Channel, or DVB
- NG SDH/SONET functionalities such as GFP, VCAT and LCAS
- optical interfaces from STM-0/STS-1 to STM-64/OC-192

11.3.1.2 Multiservice Transport Platform

A *Multiservice Transport Platform* (MSTP) can be defined as a MSPP with DWDM functions to drop selected wavelengths at a site that will provide higher aggregated capacity to transport client signals. MSTP allows to integrate circuit and data services, with efficient WDM transport and wavelength switching. Typically, MSTPs are installed in the metro and core network.

11.3.1.3 Multiservice Switching Platform

A *Multiservice Switching Platform* (MSSP) is the next-generation equivalent for cross-connect, performing efficient traffic grooming and switching at STM-N/OC-M levels but also at VC level. MSSPs should support more than just data service mapping, namely true data services multiplexing and switching. MSSP is still emerging as a next-generation network element, while MSPP and MSTP are quite mature.

11.4 Core Transport Services

Today’s telecommunications services are based on a diverse combination of technologies such as Ethernet, leased lines, IP, ATM, etc. Many of these technologies have always been clients of SDH, and others can probably also be when they need to extend their service range to wider areas.

Channelized networks organized in *n x* 64-kbit/s circuits like POTS, ISDN, or GSM, have been mapped efficiently into SDH/SONET containers in a natural way, because all of them are circuit-oriented, just like SDH/SONET. More difficult has been to match the transport of data packets and best-effort technologies like Ethernet, IP, or DVB, because the use of statistical multiplexing makes traffic variable and unpredictable. This is the opposite of SDH/SONET, which is constant and predictable, because it is based on *Time-Division Multiplexing* (TDM).

11.4.1 Next-Generation SDH

Among all the technologies that moved towards the NG SDH, Ethernet is the most remarkable. Ethernet, the standard technology for *Local Area Networks* (LANs), is cheap, easy to use, well-known, and always evolving toward higher rates. Now, when it is also being considered for access and metro networks, carriers have started to look at SDH/SONET for transporting high volumes of Ethernet traffic to get long haul transport. To make that possible SDH has needed to match the bursty and connectionless nature of Ethernet using a number of protocols:

- *Ethernet over LAPS*, defined in ITU-T X.86, is a mapping protocol of the HDLC family, which provides dynamic bit rate adaptation and frame delineation as well. It calls for contiguous concatenated bandwidth techniques (see Section 11.6) that do not match the bursty nature of Ethernet. However, what makes it inconvenient is the use of the 0x7E tag as frame delineation that forces to byte stuffing on the payload: every 0x7E occurrence is swapped by 0x7D5E. The consequence is that the frame length is data dependent and it cannot be mapped or demapped without reading the payload byte by byte.
- *Generic Framing Procedure (GFP):* defined in ITU-T Rec. G.7041. GFP is a protocol for mapping any type of data link services, including Ethernet, *Digital Video Broadcasting* (DVB) and *Storage Area Networks* (SAN)[1]. GFP provides bit rate adaptation and frame delineation. Compared with X.86, GFP has two important advantages. First, it uses a HEC-based delineation method independently of the data carried in the payload, based on a length indicator and a CRC-16 code (see Figure 11.5). Secondly, GFP uses the more efficient virtual concatenation, making GFP very popular as opposed to LAPS.
- *Virtual concatenation (VCAT)*, defined in ITU-T Rec. G.707. VCAT creates right-sized pipes for the traffic, providing granularity and compatibility with legacy SDH (see Section 11.6).
- *Link capacity adjustment scheme (LCAS)*: defined in ITU-T Rec. G.7042. LCAS allocates or de-allocates bandwidth units to match data transport requirements, or to implement additional resiliency between two transport points. VCAT can be used without LCAS, but LCAS requires VCAT (see Section 11.7).

These functions are implemented on the new MSPP nodes which are located at the edges of the network. They interact with the client data packets that are aggregated over the SDH/SONET backplane that continues unchanged. This means that the MSPPs represent the SDH Next Generation embedded in the legacy SDH network.

11.5 Generic Framing Procedure

The *Generic Framing Procedure* (GFP) is a very simple, standard protocol to map layer 2 and layer 1 signals onto SDH/SONET containers. It is point-to-point oriented and provides dynamic bit rate adaptation, delineation and framing for client data signals. Another advantage of GFP is an improved bandwidth efficiency when compared with ATM. GFP is implemented only at the edge nodes where client data interfaces are (see Figure 11.4), while the rest of the SDH/SONET network remains unchanged.

[1] SAN, as a generic acronym, can include ESCON, FICON and fiber channel as well.

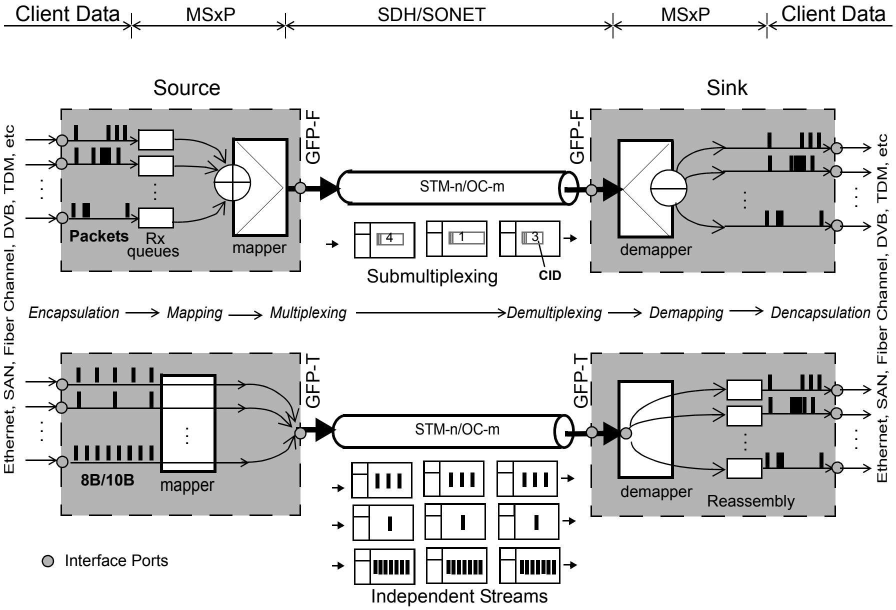
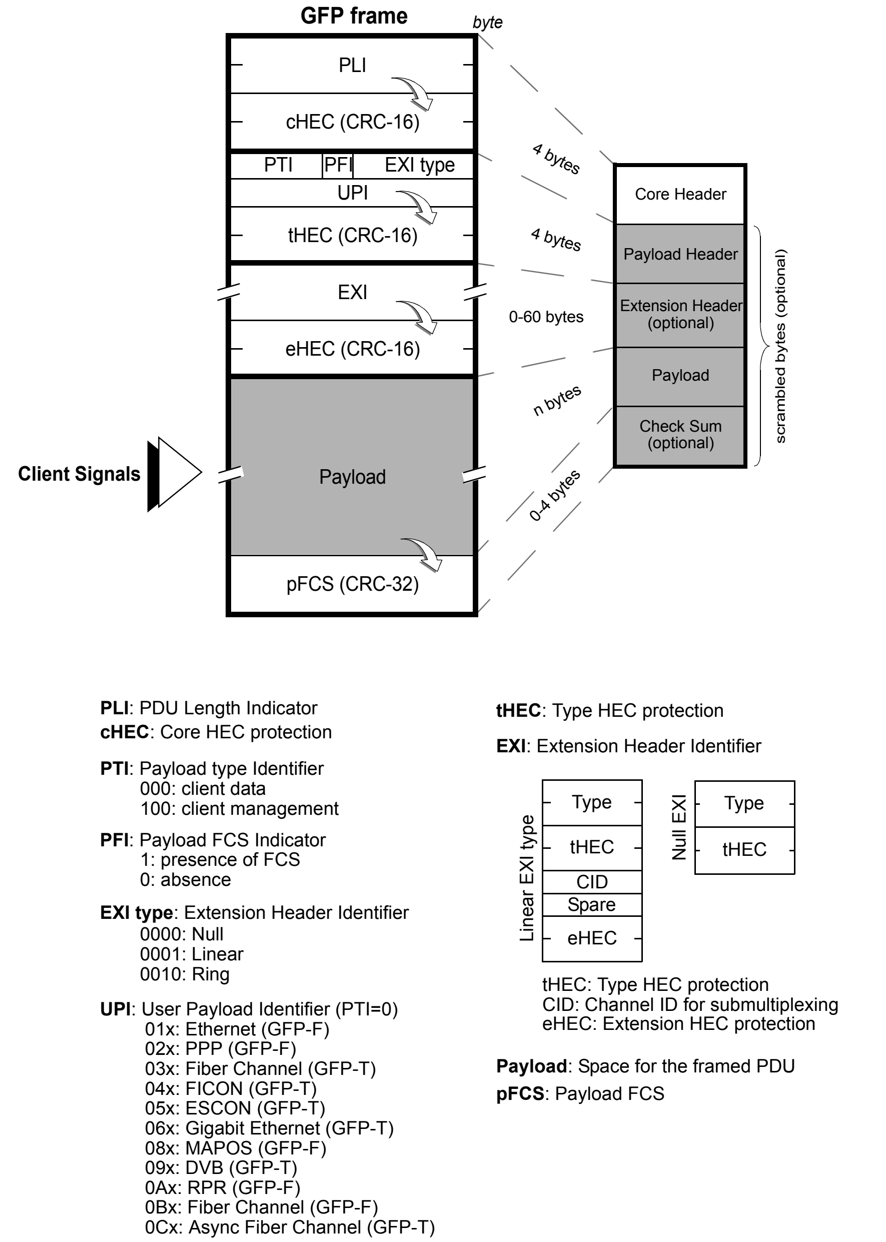


Figure 11.4 Data packet aggregation using GFP. In GFP-F packets are in queues waiting to be mapped onto a SDH channel. At the far-end packets are drop again to a queue and delivered. In GFP-T packets are encapsulated directly to the channel without waiting for the end of packet, at the far end the packet is reassembled and sent to the receiver.

GFP supports many types of protocols including those used in LAN and SAN (see Figure 11.5). GFP uses a HEC-based delineation technique similar to ATM, and it therefore does not need byte stuffing like LAPS. Then, the frame size can be easily set up to a predictable length without random increments that would happen if byte stuffing were used. The GFP HEC-based delineation has the additional capability for correcting single errors and detecting multiple errors within the core header (see Figure 11.5).

Currently, two modes of client signal adaptation are defined for GFP:

- *Frame-Mapped GFP* (GFP-F) is a layer-2 encapsulation PDU-oriented adaptation mode. It is optimized for data packet protocols (e.g. Ethernet, PPP, DVB) that are encapsulated onto variable size frames.
- *Transparent GFP* (GFP-T) is a layer-1 encapsulation or block-code oriented adaptation mode. It is optimized for protocols using 8B/10B physical layer (e.g. Fiber Channel, ESCON, 1000BASE-X, etc.) that are encapsulated onto constant size frames.



PLI: PDU Length Indicator
cHEC: Core HEC protection

PTI: Payload type Identifier
000: client data
100: client management

PFI: Payload FCS Indicator
1: presence of FCS
0: absence

EXI type: Extension Header Identifier
0000: Null
0001: Linear
0010: Ring

UPI: User Payload Identifier (PTI=0)
01x: Ethernet (GFP-F)
02x: PPP (GFP-F)
03x: Fiber Channel (GFP-T)
04x: FICON (GFP-T)
05x: ESCON (GFP-T)
06x: Gigabit Ethernet (GFP-T)
08x: MAPOS (GFP-F)
09x: DVB (GFP-T)
0Ax: RPR (GFP-F)
0Bx: Fiber Channel (GFP-F)
0Cx: Async Fiber Channel (GFP-T)

tHEC: Type HEC protection

EXI: Extension Header Identifier

tHEC: Type HEC protection
CID: Channel ID for submultiplexing
eHEC: Extension HEC protection

Payload: Space for the framed PDU
pFCS: Payload FCS

Figure 11.5 GFP frame formats and protocols. Frame delineation is done by the PLI indicating the length and cHEC which is the CRC calculated over the two octets of the PLI in a similar way as ATM does.

11.5.1 Frame–Mapped GFP

Frame–Mapped GFP (GFP-F) drops the entire client packet into a GFP frame. The encapsulation process must receive the complete client packet, then, depending on the client protocol, specific signals are removed such as certain headers, idle codes, and interframe gaps to minimize the transmission size (see Figure 11.6).

GFP-F has an optional GFP extension header (see Figure 11.5), and there are some additional fields that can be used here, such as source/destination address, port numbers, *Class of Service* (CoS), etc. The *Extension Header Identifier* (EXI) linear type supports submultiplexing onto a single channel, by means of the *Channel ID* (CID) (see Figure 11.4). This feature is intended for carriers who want to improve bandwidth usage along simple point-to-point paths by adding traffic from different low-rate sources in a single channel.

Carriers who need not only multiplexing but switching traffic as well will probably need a different approach. Traffic can be switched by means of MAC addresses or MPLS labels. These layer-2 switching capabilities are integrated in some MSPPs.

GFP and SDH/SONET is a good choice for transporting Ethernet and implementing the *Ethernet Private Line* (EPL) service defined by the *Metro Internet Forum* (MEF) in intercity or metropolitan networks.

GFP-F results in a more efficient transport, however, the encapsulation processes described above increase latency, making GFP-F inappropriate for time-sensitive protocols. This is the reason why GFP-F is used for Ethernet, PPP/IP and HDLC-like protocols where efficiency and flexibility are more important than delays.

11.5.2 Transparent GFP

Transparent GFP (GFP-T) is a protocol-independent encapsulation method in which all client signals are mapped into fixed-length GFP frames. Once the GFP frame is filled up, is transmitted immediately without waiting for the entire client data packet to be received.

GFP-T encapsulates any protocol as long as they are based on 8B/10B line coding, which is why it is often called protocol-agnostic. 8B/10B symbols and decoded, coded again to 64B/65B, and finally dropped into fixed size GFP-T frames. Therefore, it is a layer-1 mapping mechanism, because all the client characters, without exception, are transported to the far end. GFP-T is completely blind to the meaning of the codes, and it does not distinguish between information, inter-frame gaps, headers, flow control characters, overhead or idle codes.

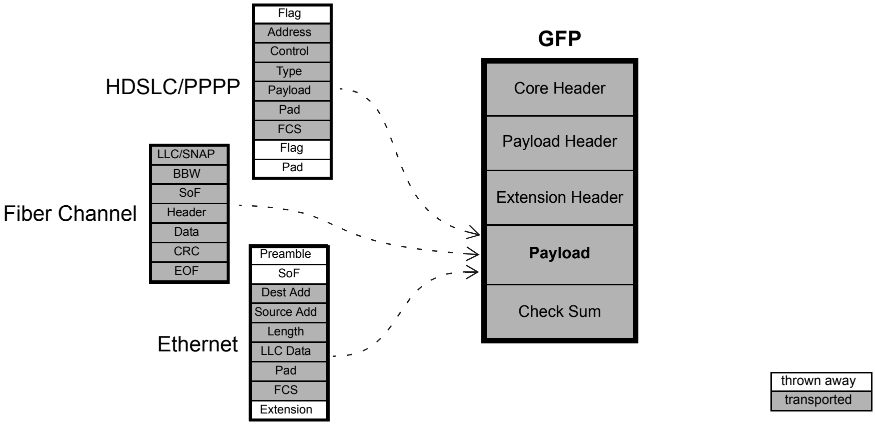


Figure 11.6 GFP mapping of client signals. Depending on the type of GFP in question, the mapping function can drop the whole signal (in the case of GFP-T), or can throw away certain delineation fields (GFP-F).

GFP-T is very good for isocronic protocols (time and delay sensitive), such as ESCON or FICON, and also for SAN. This is because it is not necessary to process client frames or to wait for arrival of the complete frame. This advantage is counteracted by loss of efficiency, because the source MSxP node still generates traffic when no data is being received from the client (see Table 11.1).

Table 11.1
Comparison between GFP-F and GFP-T modes.

Feature	*GFP-F*	*GFP-T*
Protocol transparency	low	highest
Efficiency	high	low
Isocronic or delay sensitive protocols	no	yes
Encapsulation protocol level	Layer 2 (Frames)	Layer 1 (Physical)
Optimized for	Ethernet	SAN, DVB
LCAS protection	likely	unlikely
Statistical submultiplexing of several client signals	possible	impossible
SAN transport	no	yes
Ethernet transport	optimum	possible

Advantages of statistical submultiplexing or LCAS protection are limited to variable-rate client signals. GFC-T signals are not included in this group.

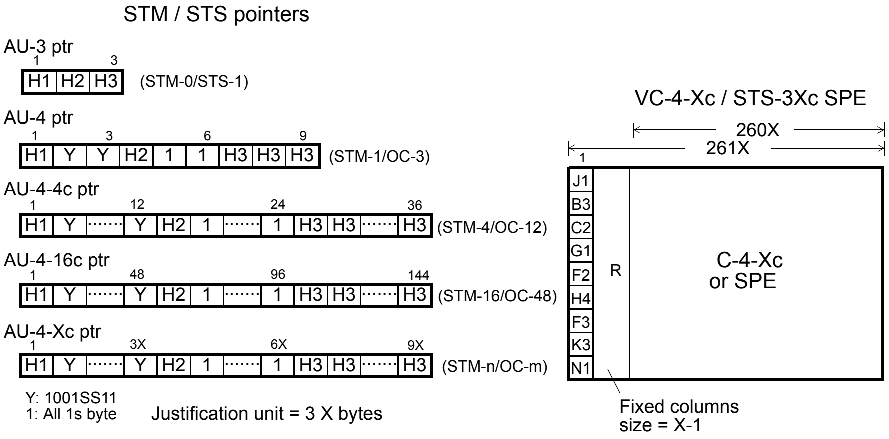


Figure 11.7 Contiguous concatenation: Pointers and containers. A VC-4-*X*c (*X* = 1, 4, 16, 64, 256) structure, where *X* represents the level. The increment/decrement unit (justification) is 3 *X*, as it depends on the level: AU-4=3 bytes, AU-4-256c=768 bytes.

11.6 CONCATENATION

Concatenation is the process of summing the bandwidth of *X* containers (C-*i*) into a larger container. This provides a bandwidth *X* times bigger than C-*i*. It is well indicated for the transport of big payloads requiring a container greater than VC-4, but it is also possible to concatenate low-capacity containers, such as VC-11, VC-12 or VC-2.

There are two concatenation methods (see Figure 11.8):

1. *Contiguous concatenation*, which creates big containers that cannot split into smaller pieces during transmission. For this, each NE must have a concatenation functionality.
2. *Virtual concatenation*, which transports the individual VCs and aggregates them at the end point of the transmission path. For this, concatenation functionality is only needed at the path termination equipment.

11.6.1 Contiguous Concatenation of VC-4

A VC-4-*X*c provides a payload area of *X* containers of C-4 type. It uses the same HO-POH used in VC-4, and with identical functionality. This structure can be transported in an STM-*n* frame (where *n* = *X*). However, other combinations are also possible; for instance, VC-4-4c can be transported in STM-16 and STM-64 frames. Concatenation guarantees the integrity of a bit sequence, because the whole container is transported as a unit across the whole network (see Table 11.2).

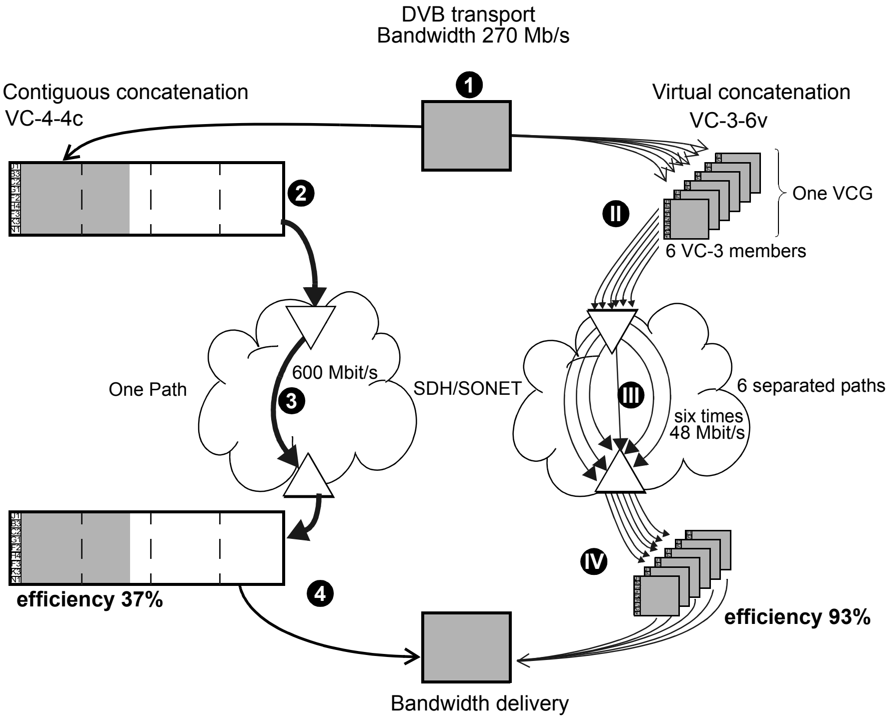


Figure 11.8 An example of contiguous concatenation and virtual concatenation. Contiguous concatenation requires support by all the nodes. Virtual concatenation allocates bandwidth more efficiently, and can be supported by legacy installations.

Obviously, an AU-4-*X*c pointer, just like any other AU pointer, indicates the position of J1, which is the first byte of the VC-4-*X*c container. The pointer takes the same value as the AU-4 pointer, while the remaining bytes take fixed values equal to Y=1001SS11 to indicate concatenation. Pointer justification is carried out the same way for all the *X* concatenated AU-4s and *X* x 3 stuffing bytes (see Figure 11.7).

Today, other more bandwidth-efficient alternatives, such as virtual concatenation, are gaining importance at the cost of contiguous concatenation.

Table 11.2
Contiguous concatenation of VC-4-Xc. X indicates the number of VC-*n*

SDH	*SONET*	*X*	*Capacity*	*Justification Unit*	*Transport*
VC-4	STS3c-SPE	1	149,760 kbit/s	3 bytes	STM-1/OC-3
VC-4-4c	STS12c-SPE	4	599,040 kbit/s	12 bytes	STM-4/OC-12
VC-4-16c	STS48c-SPE	16	2,396,160 kbit/s	48 bytes	STM-16/OC-48
VC-4-64c	STS192c-SPE	64	9,584,640 kbit/s	192 bytes	STM-64/OC-192
VC-4-256c	STS768c-SPE	256	38,338,560 kbit/s	768 bytes	STM-256/OC-768

11.6.2 Virtual Concatenation

Packet-oriented, statistically multiplexed technologies, such as IP or Ethernet, do not match well the bandwidth granularity provided by contiguous concatenation. For example, to transport 1 Gb/s, it would be necessary to allocate a VC4-16c container, which has a 2.4-Gb/s capacity. About 58% of the capacity would be wasted! (see Table 11.2).

Table 11.3
Capacity of virtually concatenated SDH VC-*n*-Xv or SONET STS-3Xv SPE

SDH	*SONET*	*Individual Capacity*	*Number (X)*	*Virtual Capacity*
VC-11	VT.15 SPE	1,600 kbit/s	1 to 64	1,600 to 102,400 kbit/s
VC-12	VT2 SPE	2,176 kbit/s	1 to 64	2,176 to 139,264 kbit/s
VC-2	VT6 SPE	6,784 kbit/s	1 to 64	6,784 to 434,176 kbit/s
VC-3	STS-1 SPE	48,384 kbit/s	1 to 256	48,384 to 12,386 kbit/s
VC-4	STS-3c SPE	149,760 kbit/s	1 to 256	149,760 to 38,338,560 kbit/s

Table 11.4
Comparison between Contiguous and Virtual Concatenation efficiency

Service	*Bit Rate*	*Contiguous Concatenation*	*Virtual Concatenation*
Ethernet	10 Mbit/s	VC-3 (20%)	VC-11-7v (89%)
Fast Ethernet	100 Mbit/s	VC-4 (67%)	VC-3-2v (99%)
Gigabit Ethernet	1000 Mbit/s	VC-4-16c (42%)	VC-4-7v (95%)
Fiber Channel	1700 Mbit/s	VC-4-16c (42%)	VC-4-12v (90%)
ATM	25 Mbit/s	VC-3 (50%)	VC-11-16c (98%)
DVB	270 Mbit/s	VC-4-4c (37%)	VC-3-6v (93%)
ESCON	160 Mbit/s	VC-4-4c (26%)	VC-3-4v (83%)

Virtual Concatenation (VCAT) is an inverse multiplexing technique that allows granular increments of bandwidth in single VC-*n* units. At the source node VCAT creates a continuous payload equivalent to *X* times the VC-n units (see Table 11.3). The set of X containers is known as a *Virtual Container Group* (VCG), and each individual VC is a *member* of the VCG. All the VC members are sent to the destination node independently, using any available path if necessary. At the destination, each VC-*n* is organized according to the indications provided by the H4 or the V5 byte, and finally delivered as a single stream to the client (see Figure 11.9).

Differential delays between VCG members are likely, because they are transported individually and may have used different paths with different latencies. Therefore, the destination node must compensate for the different delays before reassembling the payload and delivering the service.

Virtual concatenation is required only at edge nodes, and it is compatible with legacy SDH networks, despite the fact that they do not support any concatenation. To get the full benefit of VCAT, individual containers should be transported by different routes across the network, so if a link or a node fails, the connection is only partially affected. This is also a way of providing a resilience service (see Figure 11.9).

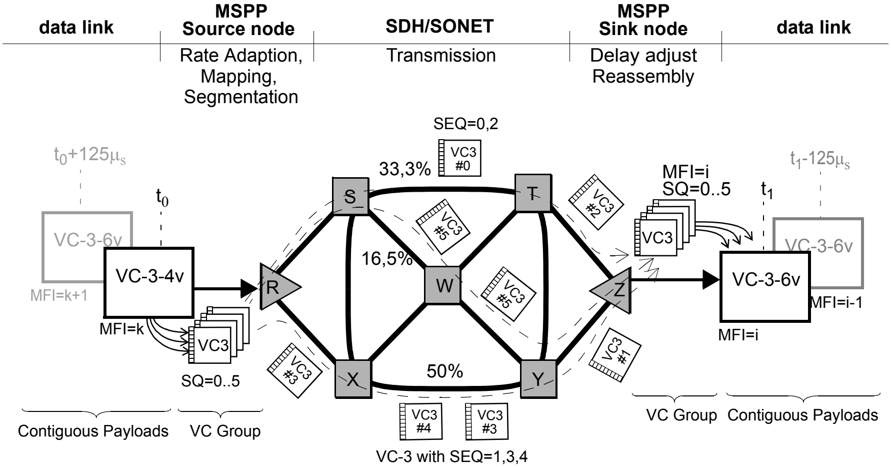


Figure 11.9 Virtual concatenation uses bandwidth more efficiently. Individual VC-3s are routed across different paths on the network. If a path fails, only a part of the bandwidth is affected and the payload is rerouted through remaining paths.

11.6.2.1 Higher-Order Virtual Concatenation

Higher-Order Virtual Concatenation (HO-VCAT) uses X times VC3 or VC4 containers (VC3/4-Xv, X = 1... 256), providing a payload capacity of X times 48,384 or 149,760 kbit/s.

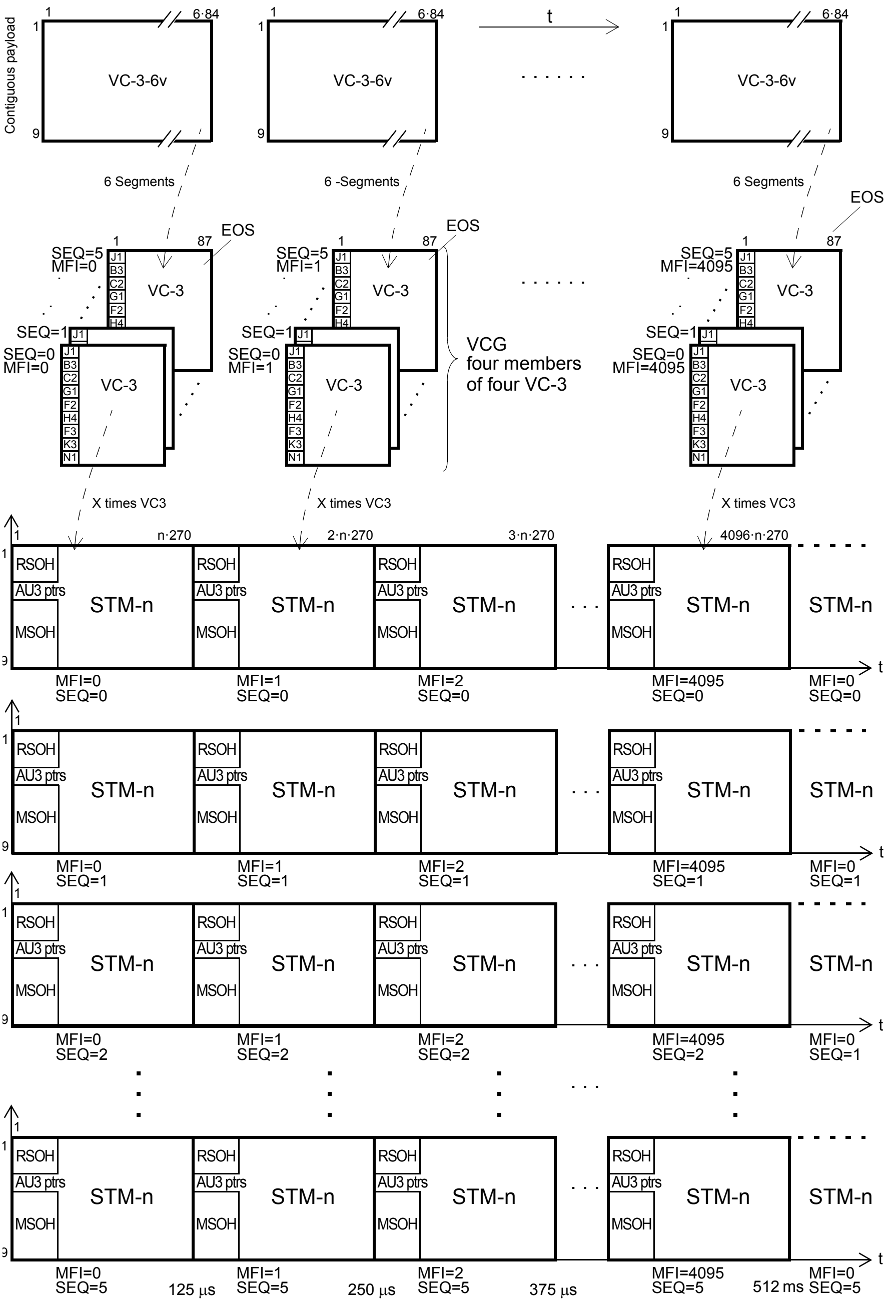


Figure 11.10 Graphical representation at the transmission (source) side of a virtual concatenation using VC-3-4v (X=4). Includes sequence (SQ), multiframe indicator (MFI), envelopes and timings of the four VC3 containers.

The virtual concatenated container VC-3/4-Xv is mapped in independent VC-3 or VC-4 envelopes that are transported individually through the network. Delays could occur between the individual VCs, and this obviously has to be compensated for when the original payload is reassembled (see Figure 11.10). A multiframe mechanism has been implemented in H4 to compensate for differential delays of up to 256 ms:

- Every individual VC has a H4 *Multiframe Indicator* (MFI) that denotes the virtual container they belong to.
- The VC also traces its position X in the VCG using the SQ number which is carried in H4.

The H4 POH byte is used for the virtual-concatenation-specific sequence and multiframe indication (see Figure 11.11).

11.6.2.2 Lower-Order Virtual Concatenation

Lower-Order Virtual Concatenation (LO-VCAT) uses X times VC11, VC12, or VC2 containers (VC11/12/2-Xv, X = 1... 64).

A VCG built with V11, VC12 or VC2 members provides a payload of X containers C11, C12 or C2; that is, a capacity of X times 1600, 2176 or 6784 kbit/s. VCG members are transported individually through the network, therefore differential delays could occur between the individual components of a VCG, that will be compensated for at the destination node before reassembling the original continuous payload (see Figure 11.9).

A multiframe mechanism has been implemented in bit 2 of K4. It includes a *Sequence Number* (SQ) and a *Multiframe Indicator* (MFI), both enable the reordering of the VCG members. The MSxP destination node will wait until the last member arrives and then compensate for delays up to 256 ms. It is important to note that K4 is a multiframe itself, received every 500 μs, and the whole multiframe sequence is repeated every 512 ms. (see Figure 11.11)

11.6.3 VCAT Setup

When installing or maintaining VCAT, it is important to carry out a number of tests to verify not only the performance of the whole virtual concatenation, but also of every single member of the VCG (see Figure 11.10). For reassembling the original client data, all the members of the VCG must arrive at the far end, keeping the delay between the first and the last member of a VCG below 256 ms. A missing member prevents the reconstruction of the payload, and if the problem persists, it causes

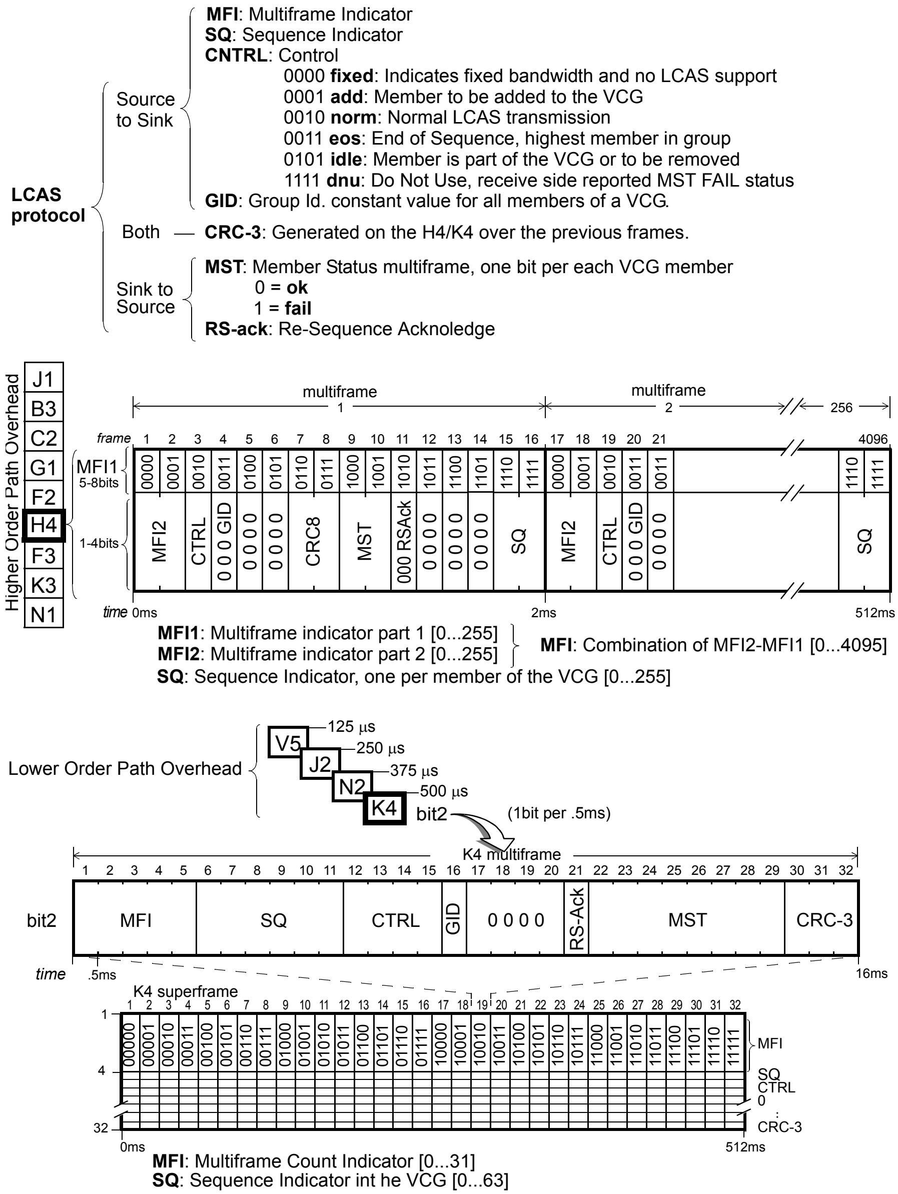


Figure 11.11 H4 and K4 codification of Multiframes. SEQ identifies each member of the group, MFI allows the differential delay calculation. H4 is part of the HO-POH overhead, which is repeated every 125 ms, so the 16-byte multiframes takes 16 ms. A complete multiframe of 4096 bytes takes 512 ms to repeat (125x4096=512 ms). K4 is part of the LO-POH overhead and is repeated every 500 ms. 32 bits are sent in a complete multiframe which takes 16ms to repeat. (500 x 32=16 ms). The bit-2 superframe is made up of a sequence of 32 multiframes and takes 512 ms to repeat.

a fault that would require reconfiguring the VCAT pipe. Additionally, jitter and wander on individual paths can cause anomalies (errors) in the transport service.

BER, latency, and event tests should verify the capacity of the network to provide the service. The VCAT granularity capacity has to be checked as well, by adding/removing members. To verify the reassembly operation, it is necessary to use a tester with the capability to insert differential delays in individual members of a VC.

11.7 LINK CAPACITY ADJUSTMENT SCHEME

The *Link Capacity Adjustment Scheme* (LCAS), standardized by the ITU-T as G.7042, was designed to manage the bandwidth allocation of a VCAT path.

What LCAS can do will be explained in the following paragraphs. However, what LCAS *cannot* do is to adapt the size of the VCAT channel according to the traffic pattern at any time. This would require direct interaction between the LCAS node and the control plane of the network, and it is not possible yet.

11.7.1 LCAS Protocol

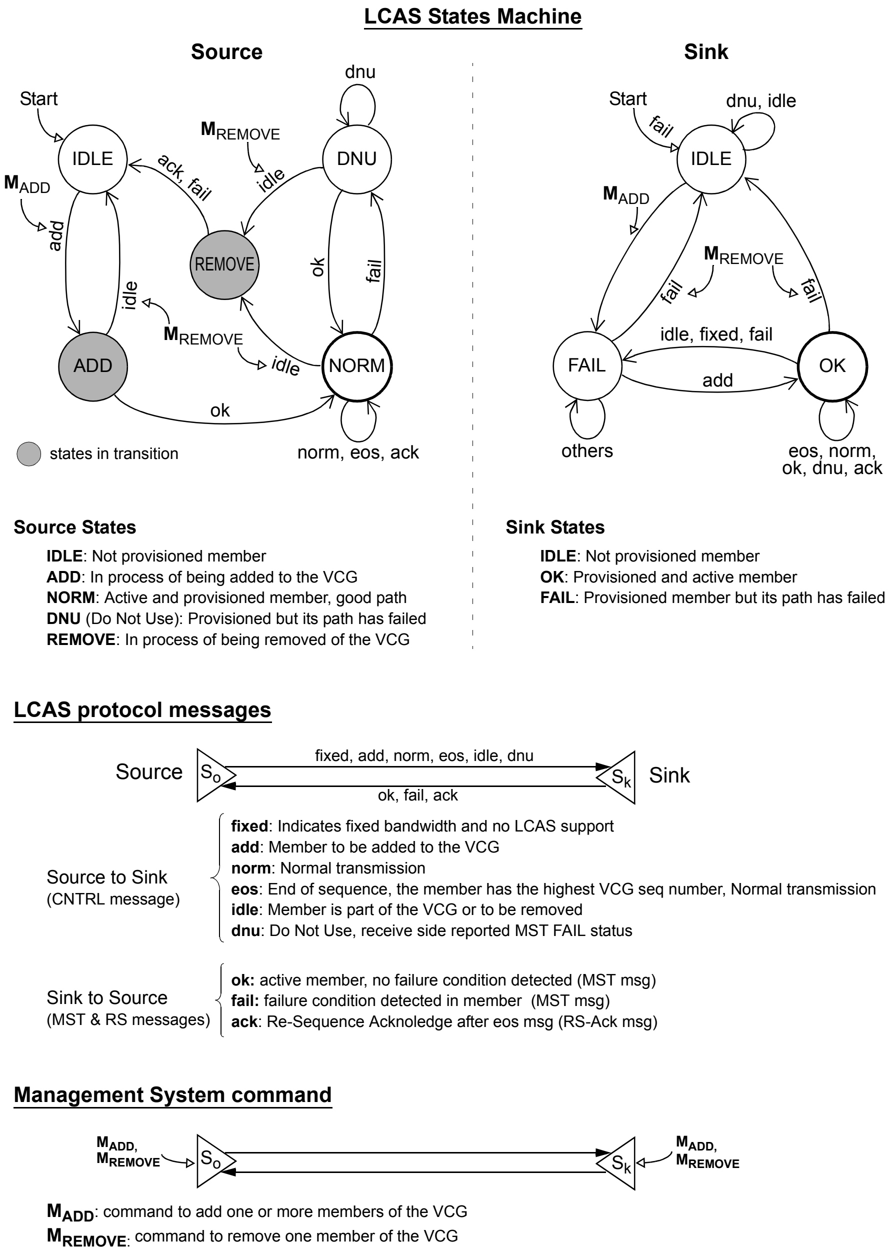
The LCAS protocol can add and remove members of a VCG[1], controlling the bandwidth of a live VCAT channel. Between the source and the sink LCAS is executed, monitoring member status and changing VCAT bandwidth use. LCAS messages are embedded in every VC member of the group providing multiple redundancy[2].

Using messages embedded in H4 or K4 (see Figure 11.11) LCAS establishes a protocol between the source node and the sink node (see Figure 11.14 and Figure 11.12) to control the concatenated group. Control messages between source to sink are a point-to-point communication. From source to sink the following information is delivered:

- *Multi-Frame Indicator* (MFI) keeps the multiframe sequence.
- *Sequence Indicator* (SQ) indicates member's sequence to reassemble correctly the client signal that was split and sent through several paths.

[1] This means that LCAS needs VCAT to work. However, VCAT does not necessarily need LCAS.

[2] It is of key importance to realise that all LCAS messages from sink to source are replicated over each member path of the VCG. That is not true for the messages from source to sink, which are specific for each member. The sink-to-source redundancy enables the management of all member seven under a severe failure state.



Source States

IDLE: Not provisioned member
ADD: In process of being added to the VCG
NORM: Active and provisioned member, good path
DNU (Do Not Use): Provisioned but its path has failed
REMOVE: In process of being removed of the VCG

Sink States

IDLE: Not provisioned member
OK: Provisioned and active member
FAIL: Provisioned member but its path has failed

LCAS protocol messages

Source to Sink (CNTRL message)

fixed: Indicates fixed bandwidth and no LCAS support
add: Member to be added to the VCG
norm: Normal transmission
eos: End of sequence, the member has the highest VCG seq number, Normal transmission
idle: Member is part of the VCG or to be removed
dnu: Do Not Use, receive side reported MST FAIL status

Sink to Source (MST & RS messages)

ok: active member, no failure condition detected (MST msg)
fail: failure condition detected in member (MST msg)
ack: Re-Sequence Acknoledge after eos msg (RS-Ack msg)

Management System command

M_{ADD}: command to add one or more members of the VCG
M_{REMOVE}: command to remove one member of the VCG

Figure 11.12 Simplified LCAS Source and Sink State Machine. ITU-T rec. G.7042 can be confusing because the same IDs can be either a state, a command or a message. To minimize that we have used lower case for protocol messages, uppercase for states, and subscript for management commands (M_{xx}).

- *Control* (CTRL) contains the LCAS protocol messages from source to sink. Possible messages are *fixed, add, norm, eos, idle,* and *dnu.*
- *Group Identification* (GID) is a constant value for all members of a VCG.

Control messages from sink to source include:

- *Member Status* (MST), which indicates to source each member status: *fail* or *OK.*
- *Re-Sequence Acknowledge* (RS-Ack) is an acknowledgment of renumbering of a sequence when a new member has been selected as *eos.*

With these control messages LCAS is also able to modify the pipe size of a VCAT channel, for example eliminating a member of the VCG that arrives with an unacceptable bit error rate, or does not arrive at all. When a capacity adjustment happens the nodes restore channel capacity in a few milliseconds.

11.7.2 Light over LCAS

It is important to note that LCAS alone cannot automatically provision dynamic bandwidth depending on the traffic demand. It cannot compute, nor establish routes to take advantage of virtual concatenation. That would be really complex and would require not only a comprehensive analysis of the traffic, but also the control of SDH/SONET architecture to set up and clear up circuits. That is far beyond the capabilities of LCAS. Route provisioning is in the hands of the *Network Management System* (NMS) (see Figure 11.12).

So, LCAS has been designed to help network operators to efficiently control NG SDH connections established at VCAT sites. The use of LCAS is not compulsory, but improves VCAT management. LCAS monitors a VCAT channel, removing or adding members to the VCG, for example after a failure detection. When at the sink side LCAS detects a failure, it sends a message to the source to remove affected member(s) of the group (see Figure 11.14). LCAS also manages the protocol to increase or decrease VCGs when it receives an indication from the NMS.

Consequently, LCAS operates inside of the core network without any interface at all to the external world (see Figure 11.13).

11.7.2.1 LCAS Operation Example

Imagine a Gigabit Ethernet service connecting two points, CP1 and CP2 that use a VC3/4v channel[1]. The VC group of four VC3 members has been set up, from A to Z following diverse paths in our example (see Figure 11.13):

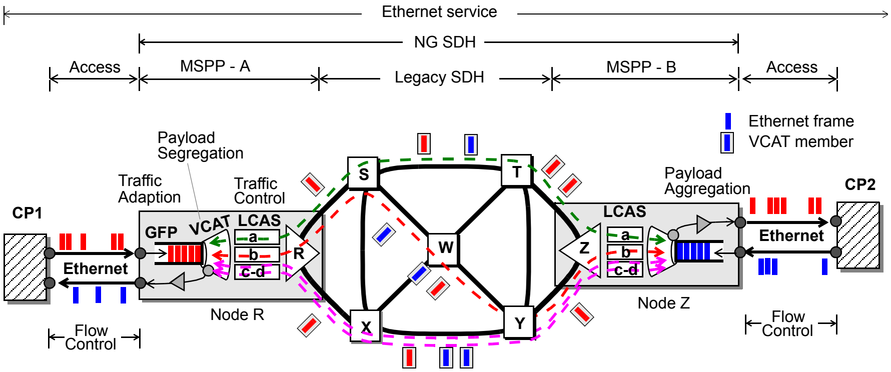


Figure 11.13 The link between node A and node Z transports Ethernet frames using a Virtual Concatenation Group of three members. Three separate LCAS protocols constantly monitor each peer connection: LCAS-a of node R talks with LCAS-a of node Z, LCAS-b(R) with LCAS-b(Z), ... LCAS-n (R) with LCAS-n (Z).

- One VC3 follows the R-S-T-Z
- One VC3 follows the R-S-W-Y-Z
- Two VC3s follows the R-X-Y-Z

Once started, the LCAS Source will send the message *add* to all members, to indicate their participation in the VCAT pipe (see Figure 11.14).

The sink has two key messages: *MST* and *RS-ACK*. The MST message can be *fail* or *OK*, and that is enough to manage failures, serious degradations, or changes to the pipe size. For example, a fault in the R-S-W-Y-Z route would cause a failure of the path b. The source will be notified of this fault with “MST= fail (b)”. Immediately the VCAT removes the faulty path from the VCG and stops using it (see Figure 11.14). Despite this, the service will still be up, and after a short interruption (maximum 128 ms), all the client’s data will have been rerouted to the three paths still alive. The interface queues at GFP will probably be larger, because the capacity is now just 75% of the original (see Figure 11.17).

It is interesting to realize that most of the packet technologies, like Ethernet, are best effort. A burst is followed by periods of low activity that, on average, compen-

[1] The capacity of a VC-3-4v channel is 193.5 Mb/s, that is 4 x 48.4 Mb/s (see Table 11.3).

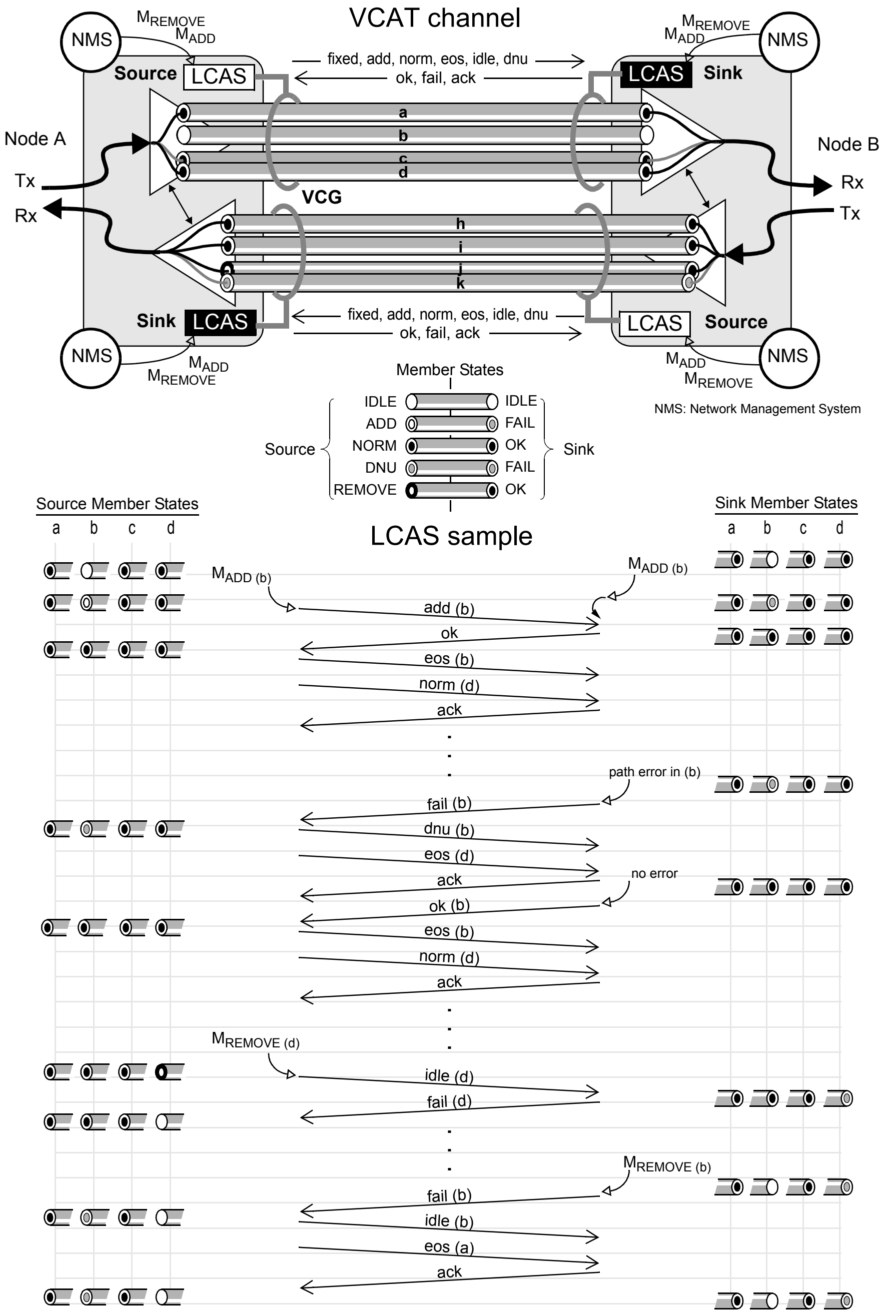


Figure 11.14 While virtual concatenation is a labelling of individual VC containers within a channel, LCAS is a two-way handshake protocol resident in H4 and K4 and executed permanently between source and sink as many times as VCAT members.

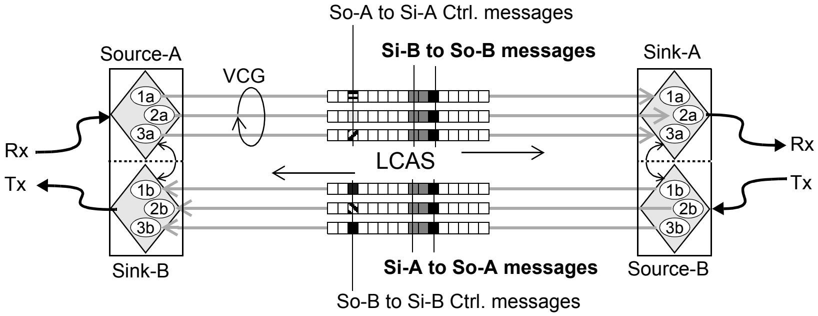


Figure 11.15 Sink messages are redundant while Source messages are unique. Notice that Sink (Si) to Source (So) messages (MST, RS-Ack) are redundant while Source to Sink are specific to each member. This means that Sink messages are repeated as many times as members in the group. It also means that the origin of sink messages is irrelevant, because all the members are sending the same information in a multi-frame. The result is a fault tolerant protocol, which is necessary when one or more members fail. This allows active members to notify the source of the situation of the whole VCG.

sate traffic peaks. So, loss of capacity at the core network can only mean larger queues at the adaption layer, and often the only consequence is more delay but not any frame loss. Many data services are very tolerant with delays so, the situation, of losing capacity, can be managed perfectly. In the case of intense traffic, MSxP can manage the flow control between the CPs, by using PAUSE protocol, in the case of Ethernet, or one equivalent for other protocols.

11.7.3 LCAS Applications

Most of the LCAS applications are related to the transport of data networks. It is of special interest in networks using GFP-F by making use of the statistical multiplexing gain.

11.7.3.1 VCAT bandwidth allocation

LCAS enables the resizing of the VCAT pipe in use when it receives an order from the NMS to increase or decrease the size. It can also automatically remove a specific VCG member, which is failing and add it again when it is recovered so that the VCAT connection is always kept alive.

11.7.3.2 Network Resilience

Resilience is probably the main LCAS application implementing a strategy known as diversification (see Figure 11.17). This strategy consists of sending traffic using

VC-4-Xv and VC-3-Xv multiframe sequence made with H4 bytes

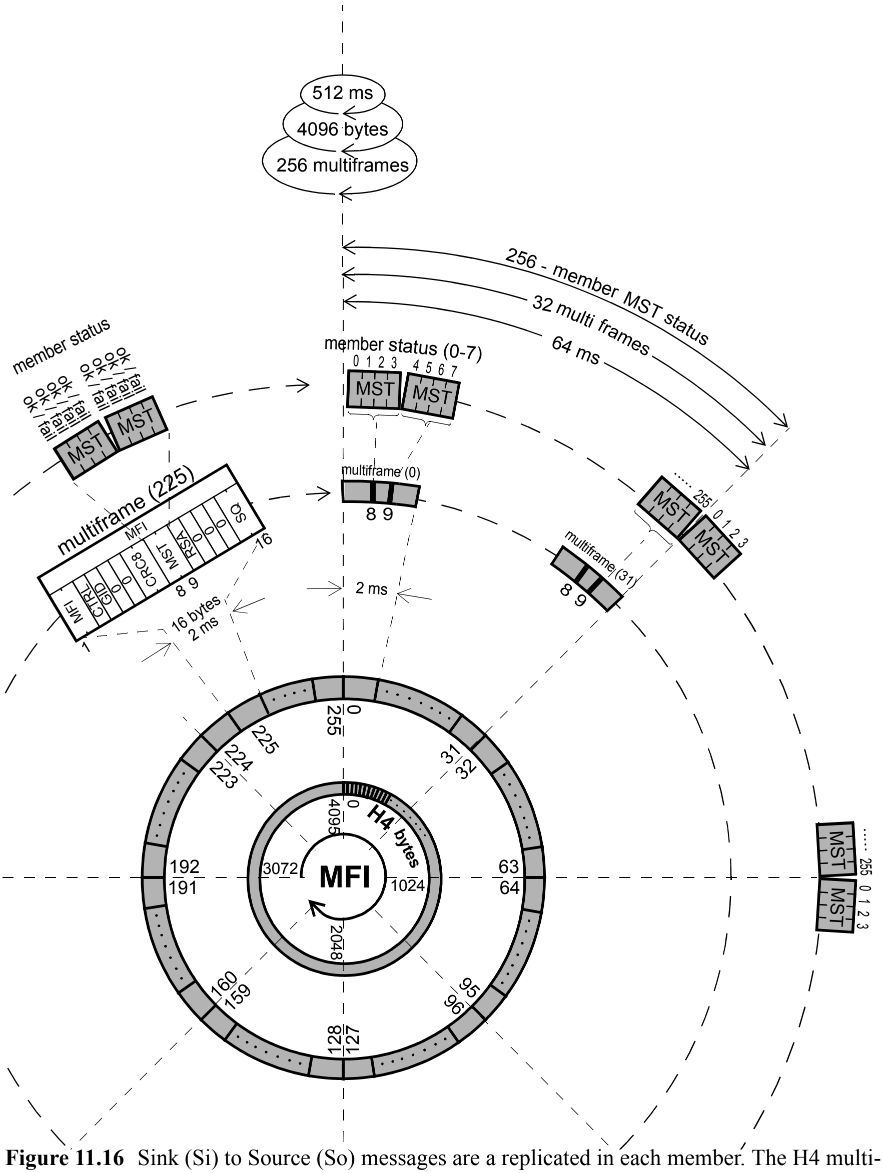


Figure 11.16 Sink (Si) to Source (So) messages are a replicated in each member. The H4 multiframe has 8 status bits therefore 32 multiframes (or 64 ms) are necessary to refresh the status of 256 members. If H4 is being used, 8 multiframes (or 128 ms) are needed to refresh up to 64 members.

several paths. In the case of a partial failure of one path, LCAS reconfigures the connection using the members still up and able to continue carrying traffic.

Diversification is especially important for packet data networks using statistical multiplexing like Ethernet. Transported signals should not be particularly sensitive to delays because the reduction of available bandwidth can increase the queues at GFP-F. LCAS restoration time runs from 64 ms for VC-4 and other higher-order virtual concatenations, and 128 ms for VC-12 and other lower-order virtual concatenations.

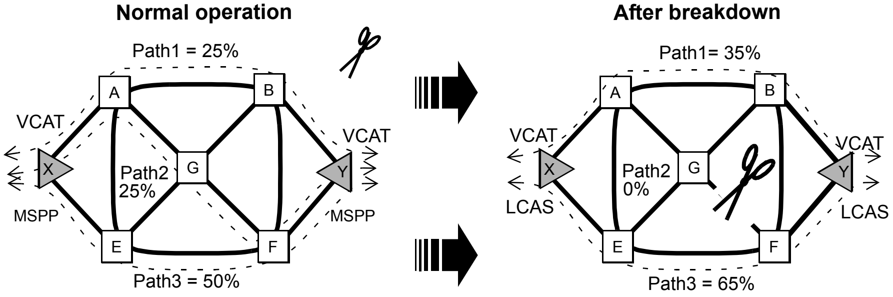


Figure 11.17 Diversification strategy between points X and Y using VCAT and LCAS.

In the case of an IP network, router topology would continue to be active, but less bandwidth would be available and consequently delay would be increased. However, constant complex configurations and reconfigurations between routers are avoided.

LCAS diversification can combine, and even replace, the existing protection architectures such as MSSPRING or MSDPRING which can also be used with NG SDH/SONET, but they are expensive, because they require spare resources which are never used except under a failure situation.

11.7.3.3 Asymmetric Configurations

It is important to note that LCAS is a unidirectional protocol that is executed independently at the two ends. This feature allows the provision of asymmetric bandwidth between two MSPP nodes to configure asymmetric links adapted to customer requirements. Asymmetric links are an interesting possibility for DSLAM connections to Internet service providers.

11.7.3.4 Cross-Domain Operation

LCAS eliminates the slow and inefficient provisioning process of legacy SDH/SONET networks. In particular, a service crosses several operators (for example international links) it is necessary to coordinate more than one configuration centre.

By using VCAT with LCAS the configuration is easier because LCAS resides only at edge nodes. The applications mentioned in the previous paragraphs, like network resilience and asymmetric links, can also be implemented in cross-domain services. It is also possible to add and remove paths from a route automatically, in real-time, and from both sides of the VCAT pipe.

11.7.4 NG SDH Event Tables

Tables summarize events and indications associated with Next Generation SDH/SONET. Ethernet events have also been included (see Table 11.5).

Table 11.5
NG SDH events and indications

SDH	*Means*	*Comments*
LOA	Loss of Alignment	Defect detected if the alignment process cannot perform the alignment of the individual VC-4s to a common multiframe start (e.g. LOA defect activated if the differential delay exceeds the size of the alignment buffer). Defined in ITU-T G.783
LOM	Loss of Multiframe	Defect declared when the OOM1 or OOM2 status is declared and the whole H4 two-stage multiframe is not recovered within m (m between 40 and 80) VC-3/4 frames. Defined in ITU-T G.783
OOM1	Out of Multiframe 1	Status declared when an error is found in the multiframe (16 frame) indication 1 (MFI1) sequence carried in bits 5 to 8 of H4 byte as defined in ITU-T G.783
OOM2	Out of Multiframe 2	Status declared when an error is found in the multiframe (256 frame) indication 2 (MFI2) sequence carried in bits 1 to 4 of H4 byte as defined in ITU-T G.783.
LFD	Loss of Frame Delineation	Loss of GFP frame delineation
cH-U	Uncorrectable cHEC error	Uncorrectable error in the GFC core header
cH-C	Correctable cHEc error	Correctable error in the GFC core header
tH-U	Uncorrectable tHEC error	Uncorrectable error in the GFC type header
tH-C	Correctable tHEC error	Correctable error in the GFC type header
eH-U	Uncorrectable eHEC error	Uncorrectable error in the GFC extension header

Table 11.5
NG SDH events and indications

eH-C	Correctable eHEC error	Correctable error found in the GFC extension header
pFCS	Payload Frame Checksum error	Error in the optional pFCS field of the GFC frame
Align	Frame alignment errors	Ethernet frame alignment error
Under	Under-sized frames	Frames with size smaller than 64 bytes with valid CRC as per 802.3 and RFC2819
Over	Over-sized frames	Frames with size greater than 1518/1522 bytes with valid CRC as per IEEE 802.3-2002 and RFC 2819
Fragm	Fragments	Frames with size smaller than 64 bytes with not valid CRC as per RFC 2819
FTL	Frames too Long	Frames with size greater than the maximum permitted as per RFC1643 and RFC2665
FCS	Frame Check Sum errors	Frames with correct size but invalid CRC

11.8 CONCLUSIONS

The technologies that today challenge the supremacy of SDH are a combination of IP with GbEthernet, RPR, and MPLS. Apparently, they can replace SDH without being compatible with the installed base. There is a very good reason behind this strategy: If 95% of the traffic is generated and terminated in IP/Ethernet, why do we need intermediate protocols?

These challenger technologies are being installed and commissioned in new metropolitan environments that provide data services. Voice and access are also on the road map. IP and Ethernet completely dominate data communications networks, the Internet, *virtual private networks* (VPNs), and LANs. They have the following key features:

- Low cost, easy installation, simple maintenance;
- Direct bandwidth provisioning;
- High flexibility in topologies;
- Scalability in terms of speed and distance.

Unfortunately, some weaknesses (poor management, lack of quality of service, jitter) still limit their application. However, they are moving in two directions:

1. Optical integration to improve performance and resilience. Several solutions have already been developed, with different levels of acceptance, including EoFiber, EoRPR, EoDWDM, and EoS.
2. IP delayering process to provide quality voice and video as well.

Originally designed for telephony services, SDH has dominated the transmission networks of the world since the early 90s, providing high-quality connections. Their key features and benefits are:

- Comprehensive OAM functions;
- Resilience mechanisms to configure fault-tolerant architectures;
- Performance monitoring and hierarchical event control;
- Synchronization, reducing jitter and wander below limits set in standards.

Today, SDH continues improving its granularity and flexibility for data transport by means of such new standards as GFP, Virtual Concatenation, and LCAS. In addition a new higher hierarchy, STM-256/OC-768, is ready to be installed.

The answer to the dilemma between the circuit-based and the packet-based approaches is a new generation of multiservice nodes providing the best of both worlds by a process of convergence (see Figure 11.2). They integrate all the remote access, routing, and switching capabilities into the same pipe. In addition, there are:

- Network interfaces: GbEthernet, STM-*n*/OC-*m*, and DWDM;
- Protocols: MPLS, ATM, IP, and VLAN;
- Client interfaces: TDM, VC, VT, Ethernet, ATM, IP, and DVB.

They maintain resilience based on protection schemes, while adding a new resilience strategy based on LCAS which is far more interesting for data packet networks. Any topology is possible, be it linear, UPSR, BLSR, or mesh. The definite proof of convergence will be if the next rate defined for Ethernet is 40 Gb/s rather than 100 Gb/s.

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