The duplexing method is likely to be based on **time-division duplexing** (**TDD**). Using TDD, instead of both the downstream and upstream signals being transmitted simultaneously by allocating each signal its own portion of the line bandwidth – as is the case with FDD – alternate time intervals are allocated for each direction of transmission. This approach means that the duration of each time interval can be varied to meet the particular configuration required. Also, since the whole of the usable frequency spectrum is used in both directions, filters are not required. As with ADSL, an active network termination is used in the customer premises.

2.6 Internet service providers

As we indicated earlier, access to the (public) Internet is through an ISP. The ISP site is located remotely from the user and, as we can conclude from the earlier sections, twisted-pair lines that make up the telephone (PSTN/ISDN) access network provide four alternative ways of communicating with an ISP:

- a pair of low bit rate modems over both access lines;
- a combination of a pair of low bit rate modems and one or more high bit rate digital circuits;
- digital circuits over both access lines;
- a pair of broadband modems.

We shall study the access protocols used with each alternative in the following subsections.

2.6.1 Home and small office users

As we show in Figure 2.30, in the case of a user at home or in a small office, access to an ISP can be implemented in two ways. In the first, access to the ISP is through two pairs of low bit rate modems: one pair to transmit the (digital) messages relating to a session/call over the user access line and a second pair to transmit the same set of messages over the called line providing the link to the ISP. Both pairs of modems have autodial/answer facilities and, at the start of each session, the subscriber/user initiates the set up of a physical path across the total network to the called (telephone) number of the ISP. Once this is in place, the user is then able to communicate with the ISP directly and, at the end of the session, clear down the physical path across the network.

The disadvantage of this method is that the use of two pairs of modems within the total path across the network means that the distance covered is double that of a single twisted-pair. Hence the effects of noise and other signal impairments are doubled and, because of this, the bit rate of each modem is limited to 33.6 kbps (V.34bis).

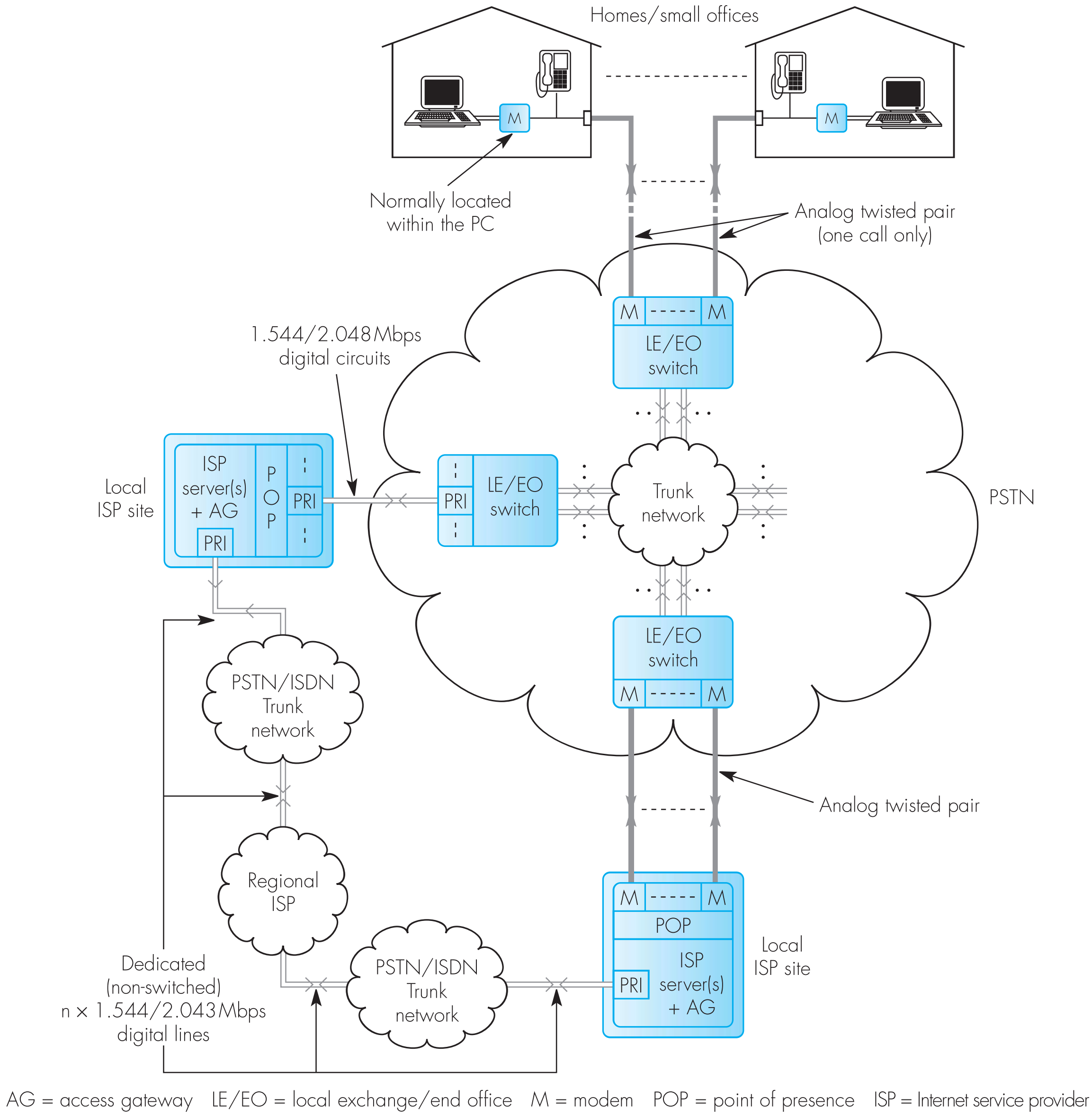


Figure 2.30 Internet access method for homes and small offices using low bit rate modems.

As we show in the figure, to overcome this restriction, instead of using a second pair of modems over each of the access lines to the ISP, a single high bit rate digital circuit is used. As we saw in Section 2.2.3, this is called a primary rate circuit, and it can support multiple independent channels each operating at up to 64 kbps. The signaling messages relating to the use of each channel for call setup and clearing are carried in a separate 64 kbps channel.

The use of a primary rate circuit in this way has two advantages. First, the removal of the second pair of modems on each access path to the ISP site means that the bit rate of the pair of modems over the user access lines can be increased to 56 kbps (V.90). Second, the physical connection to the ISP site is much simplified. A primary rate circuit provides a duplex link and the two bitstreams relating to a session are carried in the same assigned 64 kbps channel in each direction. However, since with most interactive applications a session involves, say, a short request message from the user for some information and a much larger response message from the server, to maximize the use of the available bandwidth, a V.90 modem supports a 33.6 kbps channel from the user and a 56 kbps channel in the reverse direction from the server. Each of these is then carried in the assigned 64 kbps channels.

Typically, the equipment at the ISP site comprises either a bank of low bit rate modems – one for each active subscriber/session – or one or more primary rate interface terminations. The bitstreams on the various lines/channels are then processed and broken down into the packets/messages relating to each session in an item of equipment called a **point of presence** (**POP**). The latter then relays the packets to an application – e-mail for example – running in a server at the site. The packets that need forwarding to the Internet are then passed to a second gateway that relays them out over a high bit rate circuit linking the ISP site to the nearest interior gateway – also referred to as an edge router – within the Internet. As we can see in the figure, this link is provided by a dedicated (non-switched) circuit in the PSTN/ISDN trunk network that is set up by the telcom provider and leased to the ISP. The bandwidth of the circuit is $n \times 1.544/2.048$ Mbps where n is determined by the amount of transmission bandwidth that is needed.

We outlined how this is done in Sections 2.2.4 and 2.2.5. Essentially, an add-drop multiplexer is installed at both the ISP site and the site where the (Internet) interior gateway is located. An unused circuit of the desired bandwidth is selected from the trunk cable that passes both sites. If, however, the trunk cable passes through an intermediary switching office/exchange, then the allocated high-bandwidth circuit is identified and the switch bypassed in order to form a direct link between the two sites.

2.6.2 Business users

As we indicated in the last section, a V.90 modem provides a 33.6 kbps channel from the user and a 56 kbps channel to the user. These, however, are the maximum rates, and in many instances poor quality lines mean that the actual operating rates are often lower than these. Hence, as we illustrate in Figure 2.31, many businesses choose to gain access to their ISP site using the digital circuits provide by an ISDN.

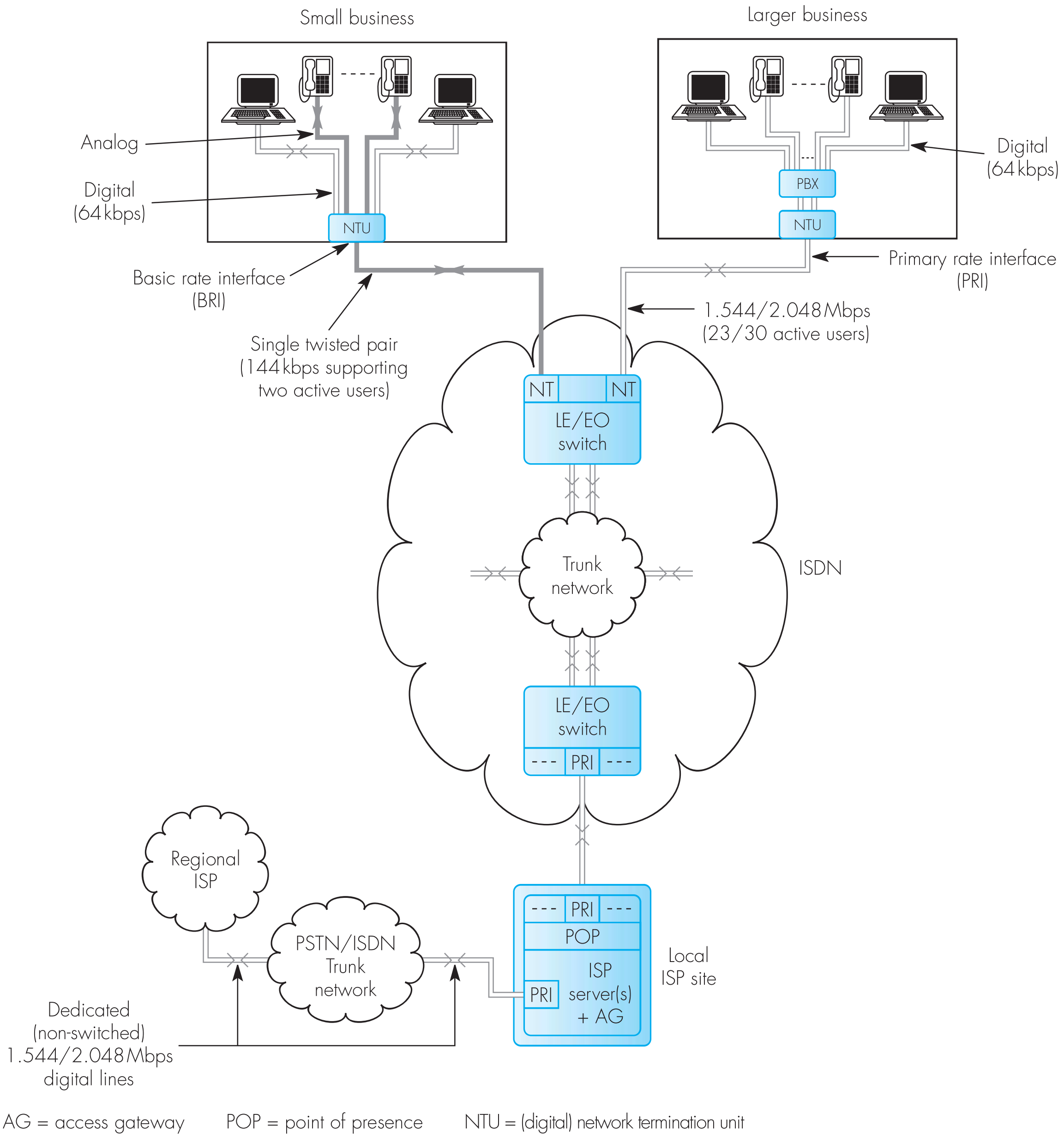


Figure 2.31 Internet access method for small and large businesses using ISDN access circuits.

In the first example, a small business is using a basic rate interface (BRI), in which, as we showed earlier in Figure 2.11, the network termination unit (NTU) provides two 64 kbps duplex channels that can be time-shared between either four or eight user devices. In the case of older analog equipment such as an analog phone, a fax machine or a PC with a modem, the analog-to-digital conversion is carried out within the NTU. All transmissions from the NTU over the subscriber line are then in a (duplex) digital form with an aggregate bit rate of 144 kbps in both directions.

In the second example, a larger business has opted to use an ISDN-enabled private branch exchange (PBX). Typically, this will support multiple terminals – phones or PCs for example – each operating at 64 kbps, or, alternatively, a number of higher bit rate channels operating at multiples of 64 kbps. The NTU in this case provides a primary rate interface (PRI) that supports either 23 or 30 active channels which can be time-shared between all the users at the site.

The signaling messages used to set up and clear all the 23/30 channels are carried over a separate 64 kbps channel. At the LE/EO, the signaling messages are then used to switch each individual user channel to the required LE/EO and from there to the ISP. Again, to support a greater number of user channels this is done using additional primary rate circuits. The functionality of the POP and server is the same as in the first example.

2.6.3 Broadband modems

We explained the operation of ADSL and VDSL modems in Section 2.5. As we saw, an ADSL modem supports both an analog telephone channel and a high bit rate digital channel. In practice, as we show in Figure 2.32, typical bit rates are up to 384 kbps from the user and up to 1.5 Mbps to the user. Hence, although the term broadband is normally used to indicate multiples of these rates, ADSL modems are also referred to as broadband modems.

The signal output by the modem at the user side is transmitted over a single twisted-pair line and is terminated at the LE/EO side by a device known as a **splitter**. This comprises two filters: a low pass (LP) filter to separate out the telephony signal and a high pass (HP) filter to separate out the high bit rate digital channel. The telephony signal is passed to the LE/EO switch in the normal way. The output of the HP filter is terminated by a piece of equipment called an ADSL **digital subscriber line access multiplexer** (**DSLAM**). This is composed of a bank of ADSL modems and a multiplexer/demultiplexer, the operation of which we described in Section 2.2.4.

The bitstreams output by each ADSL modem are then multiplexed together and transmitted to the ISP site using a leased circuit of $n \times$ 1.544/2.048 Mbps where n is the number of streams. Normally, broadband modems are *always on,* which means that a dedicated (non-switched) channel in the trunk circuit must be used to link the LE/EO to the ISP site. At the site, the incoming bitstream is first demultiplexed into separate streams. The

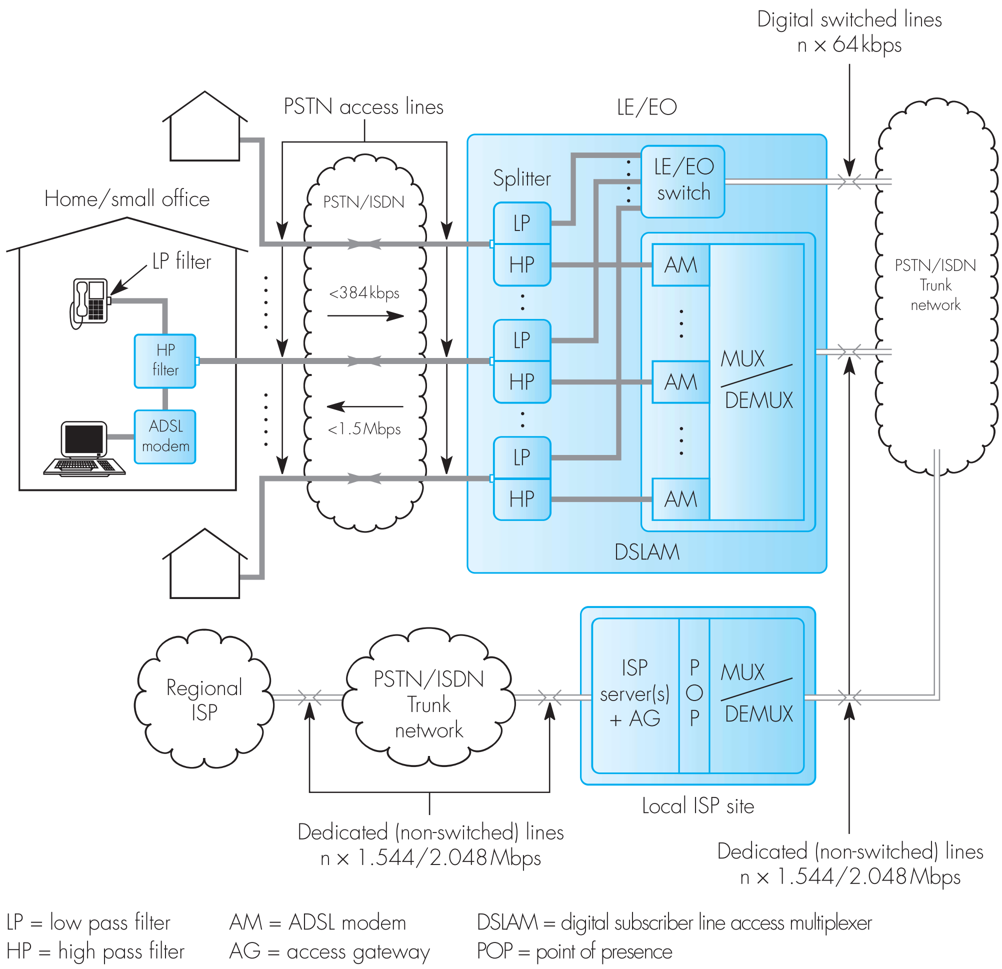


Figure 2.32 Internet access method with broadband modems.

POP then converts each stream into a stream of packets and then relays them on either to a local server at the site or to the output gateway. A similar high bit rate dedicated line is then used to link the ISP site to an interior gateway within the Internet.

The same arrangement is used with VDSL modems except a VDSL DSLAM is used. We outlined the operation of a VDSL modem in Section 2.5.2 and, as we saw, these modems operate at a much higher bit rate than ADSL. However, this is achieved only over access lines that contain a significant amount of optical fiber as we showed in Figure 2.27. Hence this restricts their use to such lines.

A final point in relation to broadband modems is that they are very easy to install and, since the telephone channel is unaffected, they provide one of the best solutions for deploying a high bandwidth link directly to the home/office.

2.6.4 The PPP link layer protocol

As we indicated earlier, the various schemes we considered in the last three sections are concerned solely with providing a direct point-to-point physical link between, say, a PC and a server within an ISP, and between the ISP access gateway and an interior gateway within the Internet. Hence a link layer protocol is needed to transfer frames of data over the various links, and an agreed network layer protocol is needed to transfer Internet packets over them.

In order to avoid the proliferation of many different protocols, the IETF has defined a standard link layer protocol to meet these requirements. This is called the **point-to-point protocol** (**PPP**) and is defined in RFC 1661/2/3 and RFC 2153. The PPP is based on the HDLC protocol we studied earlier in Section 1.4.6. To give the PPP the necessary flexibility to operate over various types of link, it has a number of features that enable it to be used in these and other applications. For example, it can operate in either the connection-oriented (reliable) or connectionless (best-effort) mode and with a variety of different types of network layer protocol. The latter feature is necessary if, for example, the protocol used over the access network link is different from the IP protocol used within the Internet.

To give it the necessary flexibility, the PPP is composed of two protocol entities: a **link control protocol** (**LCP**) and a **network control protocol** (**NCP**) In a specific application, the NCP used in both systems is selected from a set of such protocols, one for each type of network layer protocol. For example, there is an IP-NCP, an IPX-NCP and so on. The general scheme is illustrated in Figure 2.33(a) and the composition of the PPP in part (b). A state transition diagram showing the operation of the LCP and its interaction with the NCP is then illustrated in part (c). An introduction to state transition diagrams was given earlier in Section 1.4.6.

The user interface part (UIP) of the PPP is similar to that used with the LAPM protocol we showed earlier in Figure 2.23. Essentially, this issues a set of request–response service primitives that are converted by the LCP and NCP into corresponding protocol messages/PDUs. At the start of a session, the logical link between the two communicating entities – the PC and ISP for example – is in the *dead* state. On receipt of a trigger to establish a link – for example the receipt of a carrier detect signal by an ISP modem – the UIP proceeds to issue a set of service primitives first to the LCP to negotiate and establish a link with the desired options and, once this has been done, to the NCP to configure the two network layer protocols. The LCP options include whether the bit encodings are to be bit-oriented or byte-oriented, the transmission mode is to be asynchronous or synchronous, and the error control scheme is to be reliable or best-effort, all of which we described in some detail in the last chapter.

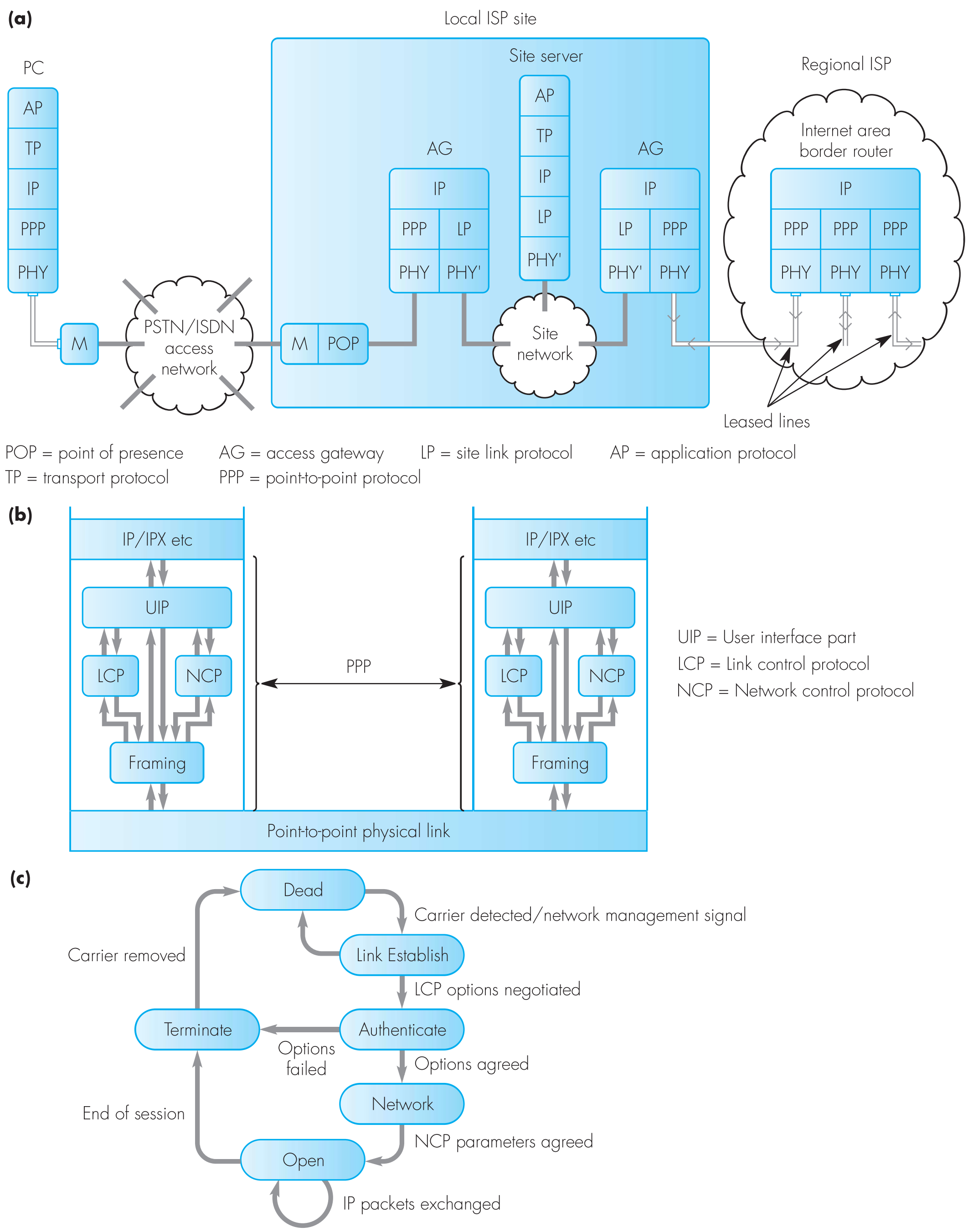


Figure 2.33 The PPP protocol: (a) the location of the PPP in relation to the other protocols; (b) the composition of the PPP; (c) a state transition diagram of the PPP.

Once the LCP options have been agreed, the link is said to be *authenticated* and the NCP is then configured. For example, in the case of a PC communicating over a switched telephone circuit with modems, the block of IP addresses allocated to the ISP is much less than the number of ISP subscribers. Hence to maximize the use of these addresses each PC only receives an IP address at the start of a session and, once the session is over, the IP address is returned to the ISP.

To do this, the NCP first determines if a free IP address is available. If not, the NCP aborts the session and the LCP terminates the link. If an IP address is available, then the NCP in the ISP sends a message containing the free IP address to the NCP in the PC. This is then passed to the IP in the network layer of the PC by the UIP. The link then enters the *open* state and the exchange of (IP) packets relating to the session – e-mail, Web browsing, etc. – can start. When the session finishes, the NCP in the PC sends a message to the NCP in the ISP to release the IP address. The LCP is then informed and this proceeds to terminate the logical link and, when this has been done, the UIP commands the modem to close down the physical link.

A similar procedure is followed for the two other cases involving switched connections. In the case of broadband modems, however, since these are always on, then the PC must be allocated a permanent IP address. Clearly, this requires a significantly larger number of IP addresses. We shall study this issue in Chapter 6 when we describe the different mechanisms that are used for allocating IP addresses in the Internet.

PPP frame format

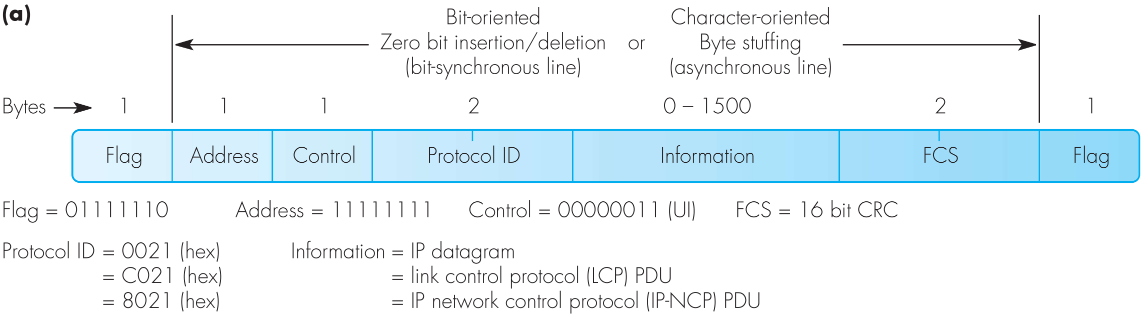
The general format of all frames used with the PPP is shown in Figure 2.34(a). Although the HDLC protocol is bit-oriented, in the PPP all frames are made up of an integral number of bytes/octets encapsulated by the opening and closing flag byte of 01111110. To achieve data transparency with the different types of synchronous transmission lines, zero bit insertion and deletion is used, the operation of which we described in Section 1.3.8.

In the case of asynchronous transmission lines, all characters are made equal to 8 bits and a form of character/byte stuffing is used, the principle of which we described in Section 1.3.3. In this application, however, in addition to flag bytes, a number of different characters/bytes are transmitted transparently. These include all the 32 control characters in the ASCII character set we identified in Figure 1.14. The list of bytes/characters and their codes is given in Figure 2.34(b).

As we can see, the escape byte/character used is 01111101 – 7D (hex) – and, whenever this is inserted, the sixth bit of the following byte/character is complemented. This rule is also used to transfer a byte/character that is equal to the escape byte.

Since the PPP is intended for use over point-to-point lines, the *address* field has no role to play. Hence it is always set to all binary 1s. Also, since in most applications all information is transferred over the line in the

(a)



(b)

Byte/character	Normal encoding	Byte-stuffed encoding
Flag	01111110	01111101 01011110
NUL	00000000	01111101 00100000
SOH	00000001	01111101 00100001
⋮		
SI	00001111	01111101 00101111
US	00011111	01111101 00111111
Escape	01111101	01111101 01011101

Figure 2.34 The point-to-point protocol (PPP): (a) frame format; (b) byte stuffing principle.

connectionless mode, as we explained in Section 1.3.8, the default value in the *control* field is 03 (hex), which is the code used to indicate an unnumbered information (UI) frame. The default length of the *protocol* ID field is two bytes. It is used to indicate the type of packet – and hence network layer protocol – that is present in the *information* field. For example, a value of 0021 (hex) is used to indicate that an IP packet/datagram is present.

As we indicated earlier, associated with the LCP are a number of messages – protocol data units (PDUs) – and, when one is present in the information field, the value in the *protocol* ID field is C021 (hex). For example, when transferring only IP datagrams over a line, an LCP *request* message can be sent to propose the use of a reduced frame header with no address or control fields and just a single byte for the protocol field. The other side then responds by returning either an *accept* message or a *reject* message in which case the default values must be used.

The default maximum size of the information field is 1500 bytes, although a different maximum size can be negotiated using LCP request–response messages. The *FCS* field is used to detect the presence of transmission errors in the frame and, as we explained in Section 1.3.8, it is based on a CRC. The FCS has a default length of 16 bits but a length of 32 bits can be negotiated using the LCP.

2.6.5 ISP networks and the Internet

As we showed in Figure 2.30 and the subsequent figures, Internet service providers (ISPs) are companies that enable users at home and in small businesses to gain access to the (public) Internet and its services. Typically, the local ISP site has a limited number of (Internet) servers that support, say, e-mail and Web access. A user at home, for example, can then communicate directly with these. However, if the addressed server is not located within the local ISP then the packet must be routed to a different site.

As we show in Figure 2.35, routing packets to their destination across the Internet can involve three levels called **tiers**. At the lowest level – **tier 3** – are the local ISP sites and campus and business LANs. In the case of a local ISP site, when an IP packet is received with a destination (IP) address that does

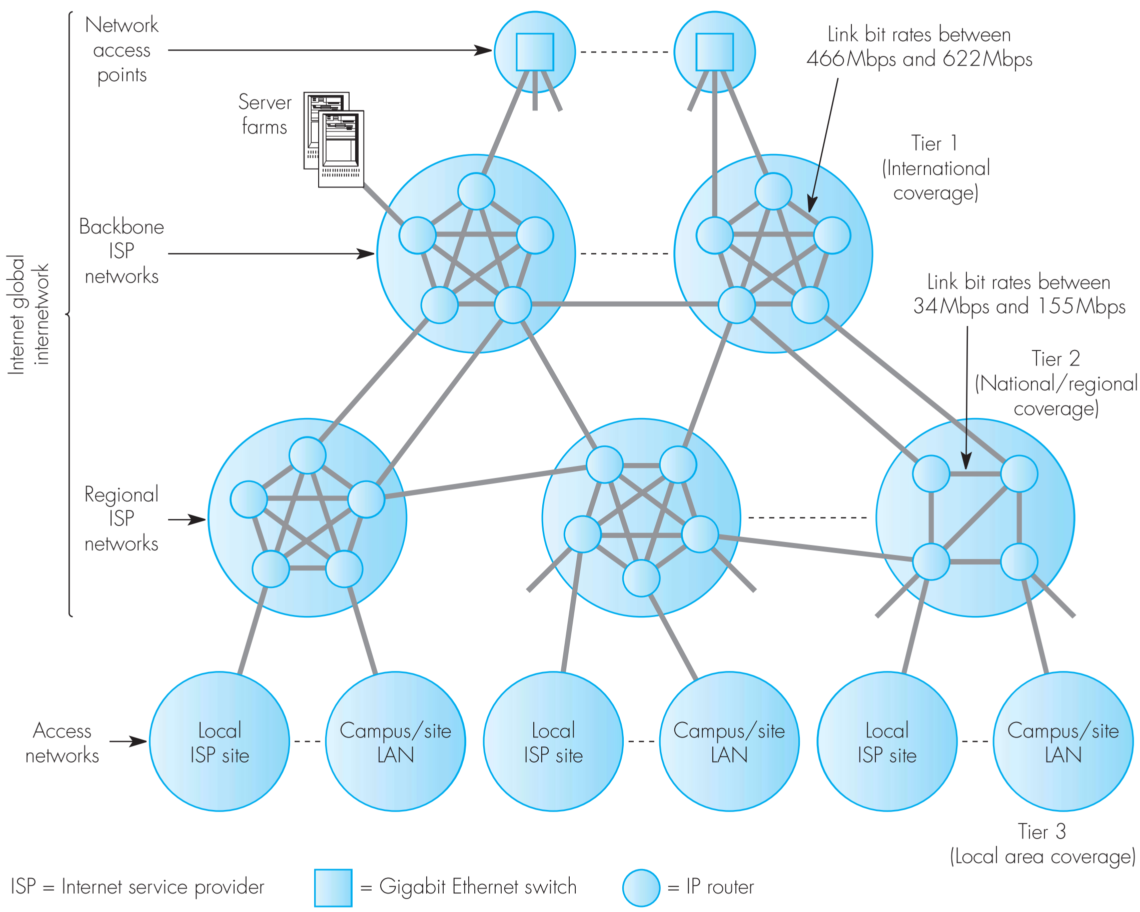


Figure 2.35 The public Internet architecture.

not belong to the set of addresses managed by the local ISP, the packet is passed directly to the router in the higher-level **regional ISP network** to which the local ISP is connected. Regional ISP networks – **tier 2** – are composed of a set of routers interconnected by high bandwidth leased lines. Each router serves a number of local ISPs, site LANs, etc. within its area of coverage. Typically, this is a city and its surrounding area and the area of coverage of the whole regional network can be that of a small country or a segment of a larger country.

As we shall see later in Chapter 6 when we discuss routing within the Internet in more detail, on receipt of a packet, the router is able to use the destination IP address in the packet header to route the packet either to a different router within the same regional network or, if the packet is addressed to a different regional network, to a higher-level router in one of the **backbone ISP networks – tier 1**. Collectively, these span the world and can route packets at very high rates.

If the packet is addressed to a computer – server/PC, etc. – that is accessible by a regional network connected to the same backbone, then the packet is forwarded to this. If this is not the case, then the packet is passed to a different backbone network using one of a set of what are called **network access points** (**NAP**). Each NAP serves a number of backbone networks and the choice of NAP to use is based on the services each offers and their cost. Typically, a NAP is a gigabit Ethernet switch. As we shall see in Section 3.4.3 of the next chapter, each line connecting a router to the switch operates at 10 Gbps and the switch is able to route packets between lines at a comparable rate.

Finally, it should be mentioned before leaving this topic that the above assumes a strictly hierarchical structure. In practice, however, because the networks at the different layers are operated by a number of different companies, private agreements are reached between them – normally on cost grounds – to provide direct links between selected routers in the same peer networks. This practice is called **private peering**. Also, there are large legacy regional/national networks still in use that do not operate using the IP protocol. Hence to provide the users of these networks with access to the services of the Internet requires additional mechanisms. We shall discuss these and other related topics later in Chapter 6.

Summary

A summary of the topics discussed in this chapter is given in Figure 2.36.