Helsinki University of Technology

Networking Laboratory

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Background Material to Networking Technology Laboratory Works

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Abstract

The purpose of this work is to give some guidelines to the Networking Technology laboratory works, help the student to understand what kind of topics he is going to confront in the laboratory works and give same pointers how the laboratory works should be conducted.

Topics covered in this background material are: N-ISDN, Analysis tree and how to conduct different kind of analysis. First topic is N-ISDN, it's functionality and services it provides to the user. Then I'll explain the concept of analysis tree and how analyses are conducted.

After reading this material student should be able to answer most of the preliminary exercises. For the rest of them student should go to Networking Laboratory and use electronic library NED.

Keywords: N-ISDN, ISDN services, analysis tree.

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1 N-ISDN

Integrated Services Digital Network (ISDN) is a system of digital phone connections, which has been available for over a decade. This system allows data to be transmitted simultaneously across the world using end-to-end connectivity.

1.1 ISDN Channels

With ISDN, voice and data are carried by bearer channels (B-Channels) occupying a bandwidth of 64 Kbps. Some switched limit B-channels to a capacity of 56 Kbps. A data channel (D-Channel) handles signaling at 16 Kbps or 64 Kbps, depending on the service type.

There are two basic types of ISDN services: Basic Rate Interface (BRI) and Primary Rate Interface (PRI). BRI consists of two 64 kbps B channels and one 16 kbps D channel for a total of 144 kbps. This basic service is intended to meet the needs of most individual users.

PRI is intended for users with greater capacity requirements. In US the channel structure is 23 B channels plus one 64 kbps D channel for a total of 1536 kbps. In Europe, PRI consists of 30 B channels plus one 64 kbps D channel for a total of 1984 kbps. It is also possible to support multiple PRI lines with one 64kbps D channel using Non-Facility Associated Signaling (NFAS).

1.1.1 D-Channel

ISDN devices attach to the network using a standard physical connector and exchange a standard set of messages with the network to request service. The content of the service-request message will vary with the different services requested; an ISDN telephone for example, will request different services from the network than will an ISDN modem. All ISDN equipment, however, will use the same protocol and same set of messages. The network and user equipment exchanges all service requests and other signaling messages over the ISDN D-channel. Typically a single D-channel will provide the signaling services for a single ISDN interface. It is possible for a single ISDN device to be connected to the network with more than one ISDN interface. In this scenario, it is possible for D-channel to provide signaling information for many ISDN interfaces. This capability saves channels and equipment resources by consolidating all signaling information on one channel; it is only available on the T-carrier ISDN interface.

Although the D-channels primary function is for user-network signaling, the exchange of these signaling messages is unlikely to use all the available bandwidth. Excess time on the D-channel is available for user's packet data and the transport of packet-mode data is the secondary function of the D-channel. The excess time is considered to be great enough to allow service providers to offer user data services at rates up to 9.6 kbps on the D-channel. This is a bargain for users because the full signaling 16 kbps of the D-channel is typically available. User-network signaling messages always have priority over data packets.

1.1.2 B-Channel

Signals exchanged on the D-channel describe the characteristics of the service that user is requesting. For example, ISDN telephony may request a circuit-mode connection at 64 kbps for the support of speech application. This profile of characteristics describes what is called bearer services. Bearer services are granted by the network, allocating a circuit-mode channel between the requesting device and the destination. At the local loop, the B-channels are designated to provide this type of service.

The primary purpose of the B-channel, then, is to carry the user's voice, audio, image, data and video signals. No service requests from the user are sent on the B-channel.

The B-channel can be used for both circuit-mode and packet-mode applications. A circuit-mode connection provides a transparent user-to-user connection, allowing the connection to be specifically suited to one type of service. In the circuit mode, protocols above the physical layer (64 kbps) are defined by the ITU-T for the B-channels; each user of a B-channel is responsible for defining the protocols to be used over the connection. It is also the responsibility of the users to assure compatibility between devices connected to B-channels. Packet-mode connections support packet switching equipment using protocols such as X.25 or frame relay. The ISDN can provide either an internal packet-mode service or provide access to an existing Packet Switched Public Data Network (PSPDN) for packet service. In either case, the protocols and procedures of the PSPDN must be adhered to when requesting packet-mode service.

B-channels can be used on an on-demand basis or on a permanent basis. If a B-channel is provisioned for permanent service, no B-channel signaling is required for the operation of the B-channel. A sample application might be the provisioning of a permanent B-channel for high-speed (64-kpbs) PSPDN access or frame relay service.

The most important point to remember is the relationship between B- and D-channels. The D-channel is used to exchange the signaling messages necessary to request services on the B-channel.

1.2 Interfaces

In the U.S., the Telephone Company will be providing its BRI customers with a U interface. The U interface is a two-wire (single pair) interface from the phone switch. It supports full-duplex data transfer over a single pair of wires, therefore only a single device can be connected to a U interface. This device is called a Network Termination 1 (NT-1). The situation is different elsewhere in the world, where the Phone Company is allowed to supply the NT-1, and thereby the customer is given an S/T interface.

The NT-1 is a relatively simple device that converts the 2-wire U interfaces into the 4-wire S/T interface. The S/T interface supports multiple devices (up to 7 devices can be placed on the S/T bus) because, while it is still a full-duplex interface, there is now pair of wires for receiving data, and another for transmit data. Today, many devices have NT-1s built into their design. This has the advantage of making the devices less expensive and easier to install, but often reduces flexibility by preventing additional devices from being connected.

Technically, ISDN devices must go through an Network Termination 2 (NT-2) device, which converts the T interface into the S interface (Note: the S and T interfaces are electrically equivalent). Virtually all ISDN devices include an NT-2 in their design. The NT-2 communicates with terminal equipment, and handles the Layer 2 and 3 ISDN protocols. Devices most commonly expect either a U interface connection (these have a built-in NT-1), or an S/T interface connection.

Devices that connect to the S/T (or S) interface include ISDN capable telephones and FAX machines, video teleconferencing equipment, bridge/routers, and terminal adapters. All devices that are designed for ISDN are designated Terminal Equipment 1 (TE1). All other communication devices that are not ISDN capable, but have a POTS telephone interface (also called the R interface), including ordinary analog telephones, FAX machines, and modems, are designated Terminal Equipment 2 (TE2). A Terminal Adapters (TA) connects a TE2 to an ISDN S/T bus.

Going one step in the opposite direction takes us inside the telephone switch. Remember that the U interface connects the switch to the customer premises equipment. This local loop connection is called Line Termination (LT function). The connection to other switches within the phone network is called Exchange Termination (ET function). The LT function and the ET function communicate via the **V** interface.

Figure 1 shows all interfaces in their own places

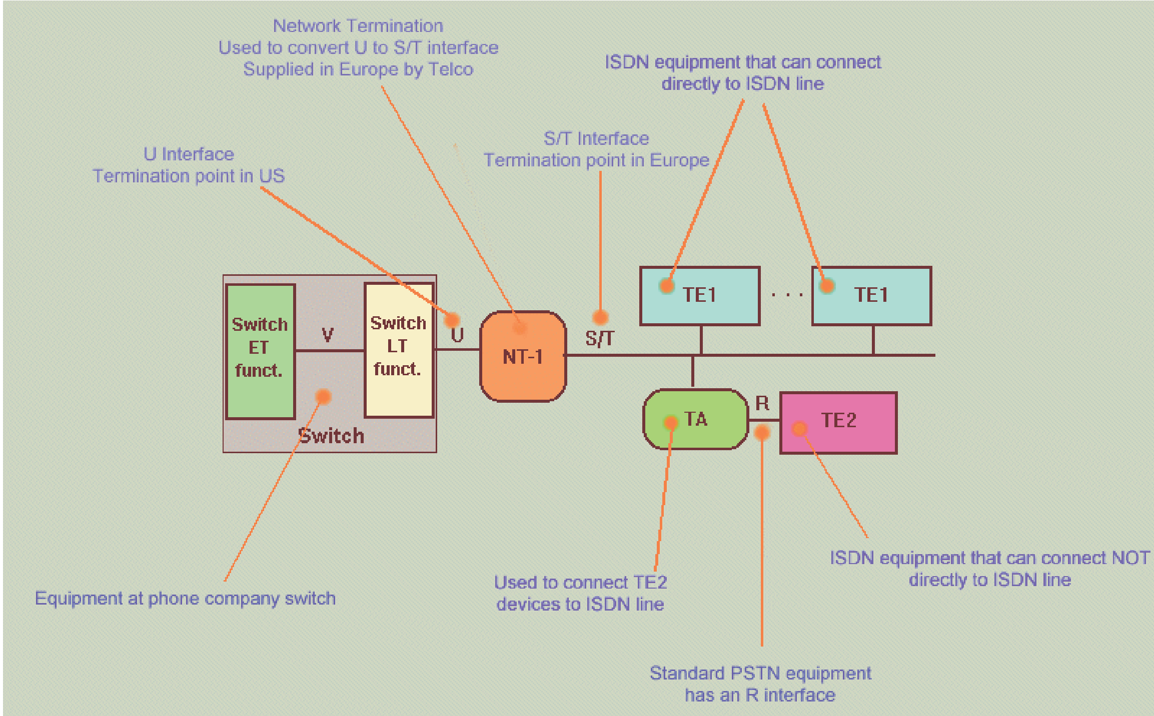

Figure 1

1.3 ISDN Layer 1 functionality

The ISDN physical layer is presented to the user at either reference point S or T (I.430). In either case, the following functions are included as physical layer functions:

- Encoding of digital data for transmission across the interface
- Full-dublex transmission of B-channel data
- Full dublex transmission of D-channel data
- Multiplexing of channels to form basic of primary access transmission structure
- Activation and deactivation of physical circuit
- Power feeding from network termination to the terminal
- Terminal identification
- Faulty terminal isolation
- D-channel contention access

The last function is needed when there is a multipoint configuration for basic access. The nature of physical interface and functionality differs for basic and primary user-network interfaces.

1.3.1 Line-coding techniques

In ISDN, both analog and digital data are transmitted using digital signals. A digital signal is a sequence of transmitted voltage pulses that is used to represent a stream of binary data. For example, a constant positive voltage level may represent binary 0, and constant negative voltage level may represent binary 1. More complex encoding schemes may be used to improve performance and quality. Used data encoding schemes are 2B1Q in North America and 4B3T in Europe.

1.3.2 Framing and multiplexing

The basic access structure consists of two 64-kpbs B-channels and one 16-kpbs channel. These channels, which produce a load of 144-kbps, are multiplexed over a 192-kbps interface at the S or T reference point. The remaining capacity is used for various framing and synchronization purposes.

As with any synchronous time –division multiplexing (TDM) scheme, basic access transmission is structured into repetitive, fixed length frames. In this case, each frame is 48 bits long; at 192-kbps, frames must repeat at a rate of one frame every 250 μsec. Figure 2 shows the frame structure; the upper frame is transmitted by the network (NT1 or NT2) to the subscriber's terminal equipment (TE); the lower frame is transmitted from the TE to the NT1 or NT2. Frame synchronization is such that each frame transmitted from a TE towards the NT is later than the frame in opposite direction by two bit-times.

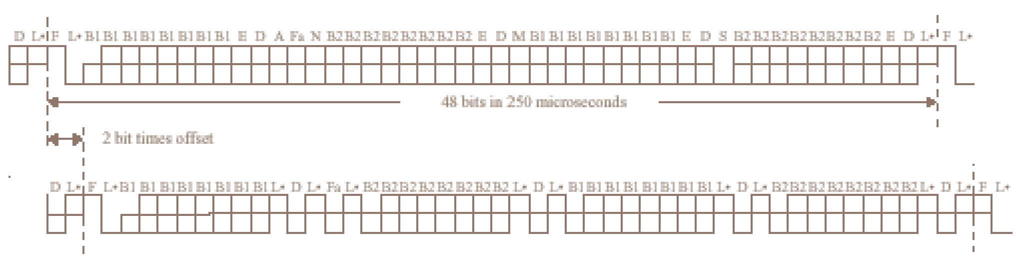

Figure 2

F = framing bit

N = bit set to a binary value N ≠ F_a (NT to TE)

L = D.C. balancing bit

B1 = bit within B-channel 1

D = D-channel bit

B2 = bit within B-channel 2

E = D-echo-cancel bit

A = bit used for activation

F_a = Auxillary framing bit

S = S-channel bit

M = multiframing bit

Each frame of 48 bits includes 16 bits from each of the two B-channels and 4-bits from the D-channel. Remaining bits have the following interpretation. Let's consider the frame structure in the TE-to-NT direction. Each frame begins with a framing bit (F) that is always transmitted as a positive pulse. This is followed by a dc balancing bit (L) that is set to a negative pulse to balance voltage. The L-F pattern thus acts to synchronize the receiver on the beginning of the frame. The specification dictates that, following these first two bit positions; the first occurrence of a zero bit will be encoded as a negative pulse. After that, the pseudoternary rule is observed. The next eight bits (B1) are from the first B-channel. This is followed by another dc balancing bit (L). Next comes a bit from the D-channel, followed by its balancing bit. This is followed by the auxiliary framing bit ($F_{a)}$, which is set to zero unless it is to be used in a multiframe structure. There follows another balancing bit (L), eight bits (B2) subsequently from the second B-channel, and another balancing bit (L). This is followed by bits from the D-channel, first B-channel, D-channel again, second B-channel, and the D-channel yet again, with each group of channel bits followed by a balancing bit.

The frame structure in the NT-to-TE direction is similar to the frame structure for transmission in the TE-to-NT direction. The following new bit replaces some of the dc balancing bits. The D-Channel echo bit (E) is a retransmission by the NT of the most recently received D bit from a retransmission by the NT of the most recently received D bit from the TE. The activation bit (A) is used to activate or deactivate a TE, allowing the device to con on line or to be placed in low power-consumption mode. The N bit is normally set to binary one. The N and M bits may be used for multiframing.

1.4 ISDN Layer 2 functionality

The primary function of the OSI Data Link Layer is to provide error-free communications link between adjacent devices. A Data link protocol has a number of specific tasks it must perform to realize this goal:

- Link connection management. Set up and terminate the data link connection between devices.
- Framing signal the beginning and end of the transmission (a frame) and delimit the user data within the frame.
- Addressing: Indicate which device is the transmitter or intended receiver of this frame
- Sequencing: Maintain sequence numbering of transmitted data frames.
- Acknowledgement: acknowledge receipt of data frames.

- Time Out: Handle those situations where an appropriate reply does not arrive within a specified time period.

- Error control: Detect bit errors, out-of-sequence frames and lost frames and correct these errors.

- Flow control: Provide a mechanism so that a receiver can avoid being flooded with data frames from a fast transmitter.

ISDN data link control procedures apply specifically to the D-channel. The ISDN data link layer protocol uses a bit-oriented protocol called LAPD.

1.4.1 Link Access Procedures on the D-Channel

The ISDN D-channel layer 2 protocol is called the Link Access Procedures on D-Channel (LAPD). LAPD defines the logical connection on the D-channel between the user device (TE or TA) and the network (NT2 or LE) across the S (or S/T) reference point or between the user (NT2) and the network (LE) across the T reference point. It supports serial, synchronous, full-dublex communication across either point-to-point or point-to-multipoint physical connection.

LAPD is officially referred to by the ITU-T as the Digital Subscriber Signaling System No. 1 (DSS 1) Data Link Layer. General principles of LAPD are described in ITU-T recommendation Q.920 (I.440) and operational procedures are described in recommendation Q.921 (I.441). LAPD is also the foundation of the protocol specified for frame relay, and frame relay-specific issues described in recommendation Q.922.

1.4.2 LAPD frames

FLAG	ADDR	CONT	INFO	FCS	FLAG

Figure 3

The unit of transmission in LAPD is a frame (Figure 3), comprising following fields:

- Flag: the bit pattern 01111110 (hex 7E). The flag indicates the beginning and end of the frame.
- Address: Identifies the user device (and protocol) that is sending, or intended to receive the frame. Always two octets.
- Control: Identifies the type of frame and may carry sequence and acknowledgement number. Either one or two octets, depending upon frame type.
- Information: Contains a DSS1 network layer message, user data or LAPD management information. Has a variable length but must be octet-aligned. This field is not present in all frames.
- FCS: Contains the 16-bit remainder from CRC calculation used to detect bit errors.

1.5 Services

ISDN user device obtains a network connection by requesting service over D-channel. When a user wants to obtain a particular service from the network, the request message over the D-channel contains a set of parameters identifying the desired service.

ISDN services are categorized based upon their scope and the source of the service. (See Figure 4) Bearer services are those that allow the user to send information from one device on the network to other. They allow information transfer and involve only lower layer functions. (I.e. layers 1 through 3 of the OSI model, depending on the service).

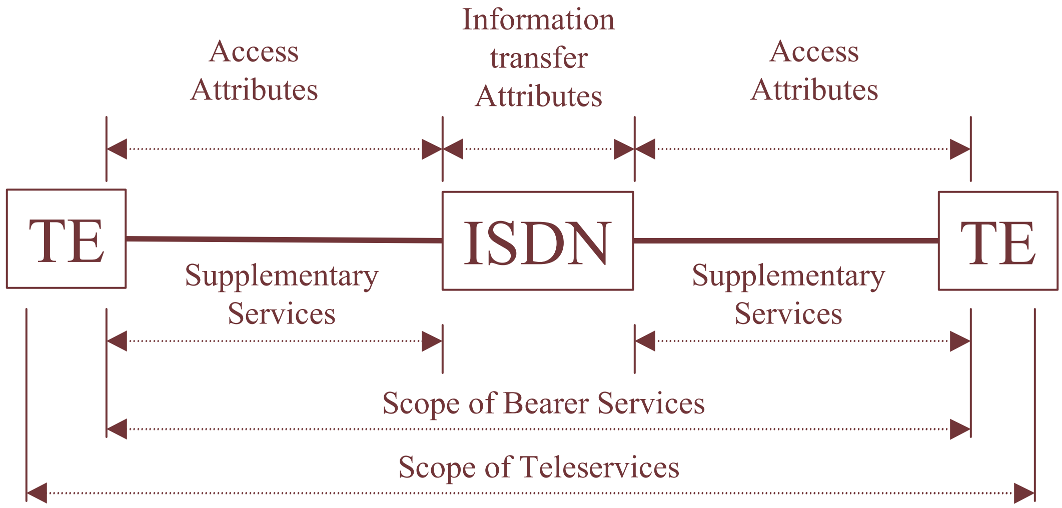

Figure 4

The users may agree between each other to use any higher-layer protocols across the requested connection. Use of higher layers is transparent to the ISDN, and the network makes no effort to assure compatibility between the higher layers. For example, the language and topic of conversation between two people talking by telephone could be considered to be higher-layer functions. The telephone network itself, however, has no knowledge of the language being spoken nor does it prevent an english-speaking customer from calling a French-speaking customer.

Teleservices are a value-added service (VAS) provided by the network, above and beyond the mere transport of bits. Teleservices can provide end-to-end communication, and they are characterized by their lower-layer attributes and higher-layer attributes. Examples of teleservices include facsimile, e-mail, videotex and electronic information system.

Two types of services are associated with the bearer and teleservices. Supplementary services are enhancements to bearer services or offer capabilities before or after the bearer service. They are considered call features or optional facilities by other networks. Supplementary services include capabilities such as call forwarding, call transfer, three-way conferencing and reverse charging. They add flexibility and enhancements to the normal bearer services. Supplementary services are typically evoked between the ISDN device and the LE but can have end-to-end significance.

1.6 Bearer Services and Attributes

Bearer services supported by an ISDN are described in ITU-T recommendation I.210 and the I.230-series recommendations. They are defined in terms of access, information transfer and general attributes. Access attributes are those characteristics describing how user accesses network functions or facilities. Information transfer attributes are those characteristics associated with the transfer of information across the network. General attributes describe other characteristics of the service, such as supplementary services, Quality-of-Service (QoS) parameters and interworking.

1.6.1 Information Transfer Attributes

Bearer services can be offered in one of three information transfer modes: circuit, frame and packet. Circuit-mode is analogous to a connection over a circuit switched network and provides a dedicated end-to-end connection for delay-sensitive applications, such as voice, audio, video and real-time data. Packet-mode is analogous to a connection via a packet switched data network. Frame-mode is similar to packet switching, except it only involves protocols up to layer2, rather that packet-mode's three-layer protocols suite.

The information transfer rate is the throughput required for the requested connection across the network. The rate is specified in thousands of bits per second for circuit mode connections or set to all zeros to indicate packet or frame mode service. The 2x64-kbps service is intended for use on the BRI and allows a user to access both B-channels on a single call.

Information transfer capability refers to the type of information that is being transferred across the network. This attribute also provides information about the requirements that the users are requesting from the network regarding the information transfer. If the network cannot meet the service requirements, request for service is denied. Unrestricted Digital Information (UDI) is a bit stream where any bit pattern may appear in an octet. UDI is the default for packet-mode transfer.

The structure attribute specifies the unit of transmission that the network will transport. Service data unit integrity implies that the frame or packet will be delivered to the destination in the same form as it was given to the network by sender.

Time slot sequence integrity (TSSI) applies to the 2x64-kbps or Nx64 transfer rate or any other service comprising an aggregate of access channels. TSSI means that information is delivered to the destination in the same relative order as submitted by the transmitter. Restricted differential time (RDTD is typically used with 8.kHz integrity - which means that all bits transmitted within a single 125 μs interval will be delivered within a corresponding 125 μs interval to the destination – to further specify that information will be delivered to receiving side within 50 ms of submission at the transmitting side. Unstructured means that no structure is implied by the given service.

The establishment of communication attribute describes when the requested service should be granted. Demand means that the requested service is needed now and the network provides the service immediately. This analogous to the way in which a dial-up connection is established over today's telephone network.

Reserved means that a user can request a service to commence at a later time and, optionally, specify the duration. Permanent establishment in analogous to a leased line. It provides service for an indefinite amount of time. The users of the service create end-to-end session using protocol-based procedures rather than on a subscription basis.

The symmetry attribute indicate whether the flow of information is one or two way and whether the speed in both directions is the same or not. Unidirectional means that transmission is one way, with no reverse communication. Bi-directional means that transmission can occur in both directions simultaneously. Bi-directional symmetry means that both users transit at the same information transfer rate, as would be expected when two telephones are connected to each other. Bi-directional asymmetric means that the requested service's information transfer rates may be different in two directions.

The communication configuration attribute describes the configuration of the connection between the end users of this service. Point-to-Point describes a service involving two users. Multipoint describes a service initiated by one user to several others, such as conference call-Broadcast would allow a single user to transmit to a group of users.

1.6.2 Access Attributes

Access attributes describes the characteristics of the connection between a user and the network. These attributes provide information about the type of channel and protocols that a user will employ to access the desired network service. Access attributes do not describe how the network should carry the user's information or how end-to-end connection should be established.

The first access attribute describes the access cannel and rate over which user access to the network should be granted. The user may request service on a N-ISDN in D-, B-channel.

There is some debate about what types of services should be available on which channel. While B-channel can be used for packet-mode transfer, many observers point out that this application is a waste of the B-channels bandwidth. Packet switching is typically used for bursty traffic; why, then, dedicate a 64-kbps channel for intermittent use? In today's environment packet-mode transfers are utilizing D- and B-channel access, and frame-mode transfers are configurable for D- and B-channels. Due to the statistical multiplexing capabilities of user devices multiple users can share the bandwidth of the channels, therefore making better utilization of the facilities.

Recall that bearer services conform to OSI layers 1 through 3. The access protocol describes which specific protocols may be used for requested service. These attributes are used by network to assure compatibility between the user and network devices. The protocols are divided into two types, signaling access and information access. Signaling access protocols refer to those used for user-network signaling, while information access protocols are used for information exchange between users. Protocols for layers 1 through 3 may be specifies for each type of access, and there are many options.

At he physical layer (OSI layer 1), ITU-T recommendations I.430 and I.431 describe the frame formats, signaling and electrical characteristics of BRI and PRI.

Recommendations I.460, I.461, V.110 (I.463, and V.120 (I.465) specify the rate adoption and multiplexing algorithms for the B-channel. Rate adoption is used to support today's non-ISDN services that do not operate at 64 kbps.

Recommendation G.711 describes PCM, and the digital voice coding scheme. Recall that to different companding algorithms are commonly used, the μ-law and A-law (used in europe). A telephone that uses one companding scheme will not be usable on a network using other scheme, so the type of companding must be indicated to network. ITU-T recommendation G.711 describes both companding schemes, as well as the conversion from one to the other.

Recommendation G.721 describes adaptive differential pulse code modulation (ADPCM), a digital voice compression algorithm. ADPCM translates 8-bit PCM sample to a 4-bit ADPCM code, meaning that voice can be transmitted at a rate of 32 kbps instead of 64 kbps.

LAPD is the Data Link Layer (OSI layer 2) protocol used on the ISDN D-channel, and it can also be used on the B-channel. LAPD is described in IT-T recommendations Q.920 (I.440) and Q.921 (I.441). X.25 LAPD may also be used for signaling or information access. ISO's HDLC protocol is another option for carrying user information. ITU-T recommendation Q.930 (I.450) and Q.931 (I.451) define the layer 3 (OSI network layer) protocol used on the D-channel to request user services. The X.25 PLP may also carry user-network signaling messages. Although the ISDN specifications allow the use of non-ISDN protocols for signaling between the user and network, all the current applications of ISDN limit the signaling protocols to LAPD and the Q.930 family protocols.

ITU-T recommendation Q.9.22 describes the link layer procedures for frame-mode and frame relay bearer services. Recommendation Q.933 describes the signaling messages to establish frame-mode services.

Finally ITU-T recommendation T.70 specifies a set of protocols supporting teletex and facsimile services; the T.70 minimal network layer protocol may be used as a layer 3 information access protocol.

1.6.3 General Attributes and Supplementary Services.

General attributes are used to further specify characteristics of an individual bearer service, such as supplementary services, quality of service, internetworking and other optional and commercial aspects of the service. Many of the general attributes remain FFS and recommendations continue to become available for these topics.

Supplementary services allow the network to provide the user with more dynamic and flexible control of how the use the network, above and beyond the mere transport of bits. Seven general categories of supplementary services have been identified by the ITU-T and described in recommendation I.250.

- Number identification, supplementary services are based on upon the presentation of one party's ISDN number to the other party
- Call offering, supplementary services are those affecting the connection and routing of call.
- Call completion, supplementary services are those affecting the completion of incoming call.
- Multiparty, supplementary services allow communications between more than a single pair of users.
- Community of interest, supplementary services allow definition of "private" networks within the public ISDN.
- Charging, supplementary services provide information about current network charges and allow the ISDN calls to be directed to a user other that the calling party.
- Additional information transfer, supplementary services allow information to be transferred between users in addition to the basic call.

These ISDN supplementary services offer capabilities to the customer than can be dynamically controlled and accessed via their CPE. In many cases, the availability of a supplementary service depends on whether it has local or end-to-end significance and the capability of the ISDN LEs to support the services and to communicate with each other.

Some of the ISDN supplementary services, such as conference calling, call forwarding and call waiting have local significance only. These services can be made available to users if the appropriate capabilities are implemented at the local ISDN LE regardless of the capabilities of the rest of the network.

The total set of supplementary services that are available from a single ISDN LE, however, may be limited unless that exchange can communicate with other ISDN switching offices. Some of the ISDN supplementary services, such as most of the number identification services and the additional information transfer service, have end-to-end significance since they require information to be carried from one user to another. An internal signaling system, must be in place to provide this interoffice communication namely SS7.

The common element with both local and end-to-end supplementary services is that the user controls their use through the signaling capabilities of the D-channel and the same user-network signaling protocol that is used to access basic ISDN services. The result is that the definition of new services only requires the extension of current protocols and has no negative impact on the existing equipment.

1.7 Teleservices

Teleservices are VAS (value added services) that may be provided to the bearer services described above. Teleservices are described in the I.240-series recommendations.

What separates teleservices from bearer services are the higher-layer, end-to-end functions (OSI layer 4 through 7). A teleservice may be offered to a user by another user of the network or by the network itself. In the former case, the ISDN merely transports the bits; in the latter case, the network is providing the VAS directly.

An example of user-to-user teleservices is the definition of the end-to-end protocols to be used for data connection during call establishment process. The added value here is that the destination can initiate the routines to handle these protocols prior to the call being accepted or possibly deny the call based on the requested protocols. An example of a user-to-network teleservices is the transport of human speech and the conversion of that speech to an analog signal at the receiving end. A more exciting network-based teleservice is the delivery of e-mail for customer or the conversion of faxes to e-mail.

2 Analysis Tree

In order to determine the destination of a call, the DX200 system must analyse the received digit sequence. This process uses data structures called analysis trees.

Each line of analysis tree specifies a destination for a given digit sequence. The maximum length of a digit sequence to be analysed is of 32 digits.

Basically there are two types of incoming calls to switch:

- Subscriber originated call (SOC)
- Trunk circuit originated call (TOC)

The selection of the tree depends on the origin of the call, i.e. on the incoming circuit group data (for TOC), or on the calling subscriber category (for SOC).

There are different trees for the calls coming from the exchange concerned and for the calls coming from exchange through the trunk circuits.

Trees 1 to 69 are standard analysis trees, reserved for a specific purpose. Trees from number 70 to 1023 can be used for connecting incoming calls to exchange.

The classification of data into several analysis trees enables the operator to define different criteria for analysis, e.g. to define a tree for a certain subscriber category, or to define a tree for international access, national access etc. The variety of trees also enables the same digit sequence coming from different circuit groups to be analysed in different trees.

3 Analysis

3.1 Priority Analysis

The priority analysis phase belongs to the so-called register analysis, which happens prior to the actual digit analysis; and it is performed for SOC only.

Priority analysis is performed to verify if the calling subscriber has a “priority” status, or in other words, to check if the “queue facility” has been assigned to the calling subscriber.

Calls coming from priority subscribers may queue for the release of a switching device, such as DTMF-receiver in the MFSU, or a free or an outgoing circuit. In the case of overload, the exchange will restrict incoming calls on the basis of the subscriber category; so that calls from ordinary subscribers are rejected and calls coming from priority subscribers will be let through.

3.2 Origin analysis

Origin analysis refers to the analysis of the calling subscriber facilities (for SOC) such as line closed, abbreviated dialing, outgoing call barring category, etc.

Another important task of the origin analysis is to select the charging origin (C_ORG) to be used in the call. This is performed for both SOC and TOC. The charging origin gives information about the calling party and it is used to determine the cost of the call as well as to keep charging records that are used between network operators,

By using different charging origin parameters, the operator can apply different charge to call directed to the same destination depending on who’s made the call.

The C_ORG parameter comes to the DX 200 system from the calling subscriber category and facilities, or from the basic data of incoming circuit.

3.3 Dialing pre-analysis

If the calling subscriber has used facility code, the dialing pre-analysis is needed in order to translate the dialing code (*, #) into the actual B-digits. This is performed for SOC only.

3.4 End-of-dialing analysis

End-of-dialing analysis is needed to verify the facilities of the called subscriber, such as call diversion, incoming call barring, etc. If for instance, call diversion is active, the new digits will have to be re-analysed. Thus, this analysis phase is required when the called subscriber belongs to the exchange concerned.

3.5 Call barring analysis

Call barring analysis is performed if the calling subscriber has activated some barring features for his subscription (for SOC only). There can be several barring cases build up in the exchange, so that the subscriber may have barred all outgoing calls, outgoing national calls, calls to service numbers and so on.

The result of the analysis will be following:

- Continue call set-up: in case barring analysis does not satisfy the barring conditions.
- Stop call set-up: in case the barring analysis satisfies the barring conditions

3.6 Analysis tree selection

The tree selection procedure determines the basic tree where the digit analysis will be performed.

In case of a subscriber-originated call (SOC), the analysis tree is determined on the subscriber category. In the case of a trunk circuit originated call (TOC), the incoming analysis tree number is attached to the basic data of the circuit group.

3.7 Digit analysis

Broadly speaking all of the analyses so far described are also digit analyses, in the sense that at every stage digits are analysed in order to get a result. But up to now the destination of the call could not be determined, the result of one the previous analyses was the input for the next phase.

After the analysis tree selection, the task of the call control process is to perform actual analysis of the dialed digit sequence in the selected tree.

The digit analysis is performed in the central memory with the aid of several analysis files, from one record to another, digit by digit, and the final result of the digit analysis process is a destination.

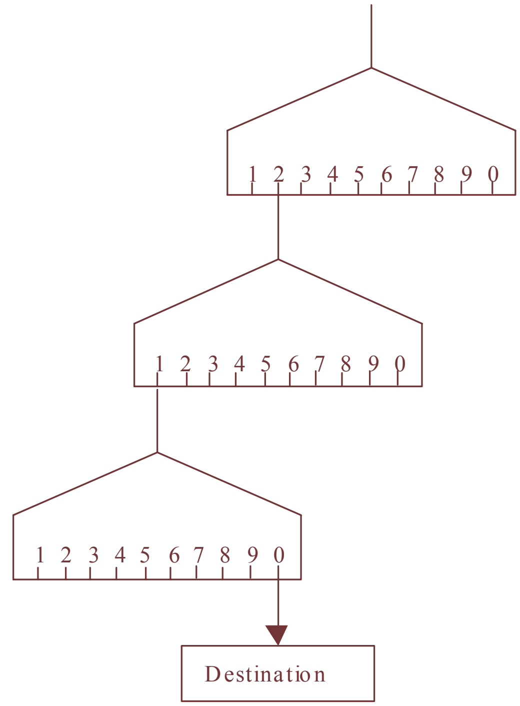

Figure 5

Attached to a destination there can be a primary subdestination (alternative 0, AL0) and four other options called alternative subdestinations (AL1 through AL4). Therefore, once the destination is obtained, the next step is to perform the "route selection" in order to choose the path to be used to route the call towards its destination. A number of different parameters influence the route selection, but basically its task is to choose the available route in order of priority (AL0 through AL4).

Selecting a physical circuit on a certain 2 MBIT PCM line, and forwarding the call ahead on that circuit does the routing of the call itself.

The exchange uses Call Control programs to supervise the call once it enters the switch. The Call Control uses digit analysis and route selection to connect the call with the help of the switching process SWICOP, which performs hunting.

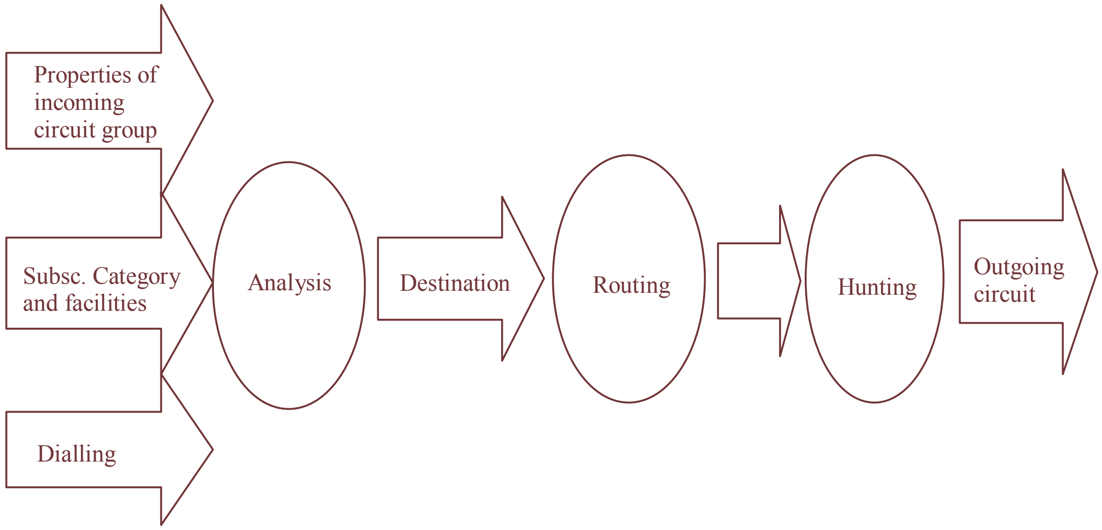

Figure 6

Figure 6 illustrates the tree main phases of the routing specification: digit analysis, route selection and circuit hunting.

4 Signaling System 7

4.1 Introduction to SS7

When you make calls using your telephone or PC with modem connection, you generally use the PSTN. You hear progress tones such as a dial tone or a busy tone that tell you that your call is being processed. These are alerting tones that are produced by the PSTN. Alerting tones are used to communicate with the PSTN and users, and telecommunications computers. Switches communicate also with each other through standards-based signaling. Signaling is the backbone for interconnection between carrier, cellular, and wireless networks. Signals are the medium that set up and tear down your calls.

A major breakthrough in signaling networks was to separate the signaling path from voice and data conversations. This is called "common channel interoffice signaling (CCS)." CCS, also known as out-of-band signaling, is a data network that overlays a carrier's switching network. Using CCS increases network intelligence, efficiency, automation, and functionality. CCS has evolved into a standard called Signaling System 7 (SS7), a protocol that lowered costs and increased network reliability even further. With SS7, all carriers are able to interoperate as a consistent and seamless network. Services such as global billing, wireless roaming and 800 number calling rely on the SS7 protocol to exchange messages reliably.

Typically, SS7 is implemented on a separate data network within the PSTN and provides call setup and teardown, network management, fault resolution, and traffic management services. The SS7 network is used solely for network control, and the only data sent over it is signaling messages.

The SS7 protocols that convey signaling information between switching systems (called signaling points) in the PSTN are carried on a special overlay network used exclusively for signaling. The signaling points use routing information in the SS7 signals to transfer calls to their final destinations.

The SS7 network features include:

- Control over the establishment of calls across the PSTN.
- Routing, billing, information exchange functions, specialized call treatments, and enhanced routing.
- Common channel signaling in which signaling information for different connections travels on separate dedicated signaling channels.
- Voice and data connections that travel on bearer channels.

The SS7 architecture consists of the following signaling points:

- Service Switching Points (SSPs) are telephone switches equipped with SS7 software and signaling links. Each SSP is connected to both STPs in a mated pair.

- Signal Transfer Points (STPs) receive and route incoming signaling messages toward their destinations. STPs are deployed in mated pairs and share the traffic between them.

- Service Control Points (SCPs) are databases that provide the necessary information for special call processing and routing, including 0800 and 0700 call services, credit card calls, local number portability, cellular roaming services, and advanced call center applications.

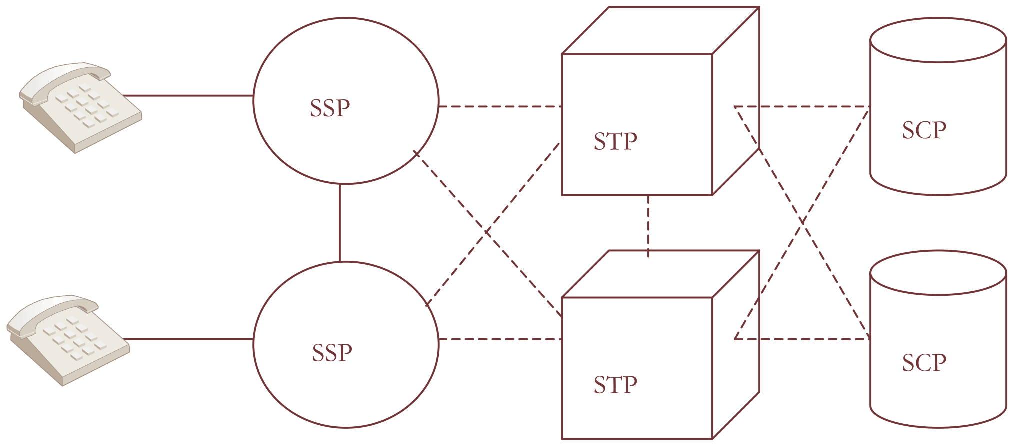

Figure 7: SS7 Signaling points

4.2 Signaling points

Each signaling point in the SS7 network is uniquely identified by a numeric point code. Point codes are carried in signaling messages exchanged between signaling points to identify the source and destination of each message. Each signaling point uses a routing table to select the appropriate signaling path for each message.

Network traffic between signaling points may be routed via a packet switch called an STP. The role of an STP is to route each incoming message to an outgoing signaling link based on routing information contained in the SS7 message. Because it acts as a network hub, an STP provides improved utilization of the SS7 network by eliminating the need for direct links between signaling points. An STP may perform global title translation, a procedure by which the destination signaling point is determined from digits present in the signaling message.

Because the SS7 network is critical to call processing, SCPs and STPs are usually deployed in mated pair configurations in separate physical locations to ensure network-wide service in the event of an isolated failure. Links between signaling points are also provisioned in pairs. Traffic is shared across all links in the link set. If one of the links fails, the signaling traffic is rerouted over another link in the link set. The SS7 protocol provides both error correction and retransmission capabilities to allow continued service in the event of signaling point or link failures.

4.3 Numbering of Signaling Network

Each network element has its own numbering for its PCM trunk lines. These numbers do not conform with the number used by the other network elements. There must be a common naming conventional, because the signaling of several speech TLS's is sent in the same signaling link.

The signaling points must be able to identify the speech time slots, which are being signaled. The time slots are also called circuits, because they act as conventional circuit switched in analog networks. The identifier, Circuit Indication Code, CIC is used to distinguish between the circuits. The circuit indicator consists of the PCM line identifier and the number of the time slot.

The elements define all signaling links between the signaling points by using common identifier. These codes are called Signaling Link Codes, SLC.

Two countries can use same signaling point codes for their signaling points. For this reason we must be able to define several signaling networks. A network element can belong simultaneously to several signaling networks and it may have different SPC's in different networks. Typically, an international traffic exchange belongs to two signaling networks, national and international, and it has two different signaling point codes, one for respective signaling networks. It is possible to have two different international signaling networks, IN0 and IN1, as well as two National signaling Networks, NA0 and NA1.

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