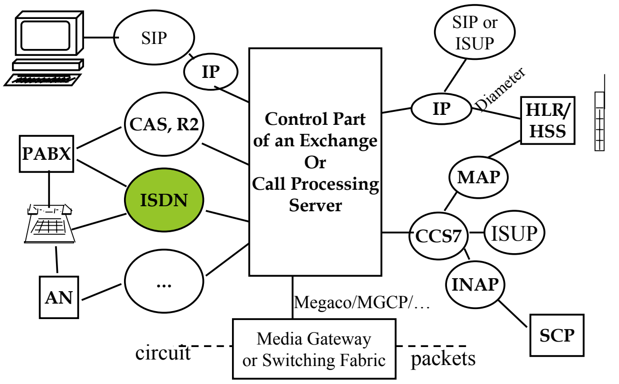
Integrated Services Digital Network

- Some repetition
- ISDN principles and the structure of ISDN access
  - structure
  - interfaces
  - physical layer
- ISDN signaling
  - bearer and telecommunication services
  - layer 1
  - layer 2
  - layer 3
- Efficiency of signaling
- ISDN evaluation and summary

Summary of course scope (1)

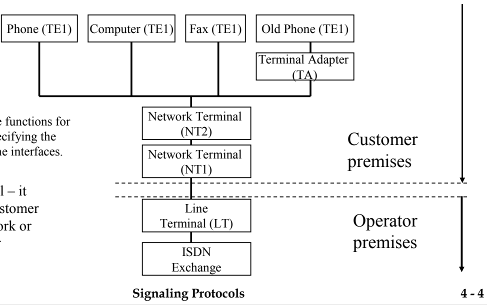


Some repetition

- Channel Associated Signaling (CAS) is tightly tied to the voice channel either in space, time or frequency -> no signaling unless voice channel is reserved.
- In in-band signaling, the voice path itself is used to carry signals.
- CAS has many limitations: in a PCM -frame one tsl needs to be dedicated to signaling and a multi-frame of 16 frames needs to be maintained. The set of signals is limited.
- Channel Associated R2-signaling is the first widely adopted, standardized CAS signaling system (but was never used in every country).
- Call setup or register signaling vs. line signaling may use different representation of signals as well as different channels to carry the signals.
- Any CAS system provides only a limited set of signals, often their semantics is context dependent.

ISDN -access has a set of standardized interfaces

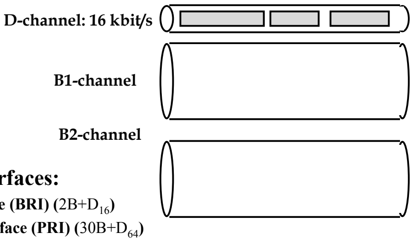
- ISDN-access provides a bus for connecting user terminals, the max of 8 terminals can be attached.
- Many interfaces are specified between *logical entities* in the access



The rectangles are functions for the purpose of specifying the functioning and the interfaces.

NT2 is optional – it represents a customer premises network or PBX or similar

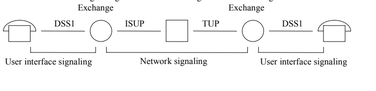
ISDN Basic Interface provides 2 x 64kbit/s to the user



- Two types of Interfaces:
  - Basic Rate Interface (BRI) (2B+D_{16})
  - Primary Rate Interface (PRI) (30B+D_{64})
- BRI provides two B-channels for information transfer and a signaling channel (D-channel) :
  - Two independent terminals can use one B-channel each at a time.
  - The main purpose of the D-channel is transport of signaling between the terminals and the local ISDN exchange. Packet mode transfer is used on the D-channel.

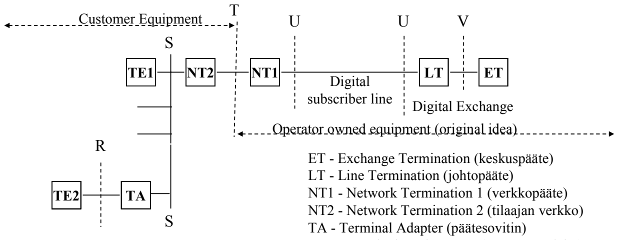
Message based signaling systems

- Message based signaling has been developed to improve the control possibilities of the network by terminals.
- Message based signaling can be used only by Computer controlled, fully digital exchanges.
- Message based signaling is natural for computers - the signaling information is largely in the same format in which it is processed and stored.
- Message based signaling is based on ITU-T:n SS6 (now CCS7 and ISDN) recommendations.



DSS1 – Digital Subscriber Signaling System No 1, ISUP – ISDN User Part, TUP – Telephony User Part
ISDN – Integrated Services Digital Network.

ISDN Access and ISDN Interfaces



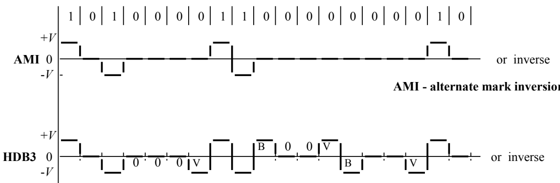
ET - Exchange Termination (keskuspääte)
LT - Line Termination (johtopääte)
NT1 - Network Termination 1 (verkkopääte)
NT2 - Network Termination 2 (tilaajan verkko)
TA - Terminal Adapter (päätesovitin)
TE1 - Terminal Equipment 1 (ISDN päätelaite)
TE2 - Terminal Equipment 2 (muu päätelaite)

2B + D (144 kbit/s)

- S, max 8 — transmission direction, power — brutto: 192 kbit/s (international standard), AMI -code
- U — brutto: 160 kbit/s (de facto or national standard), defacto: 2B1Q -code

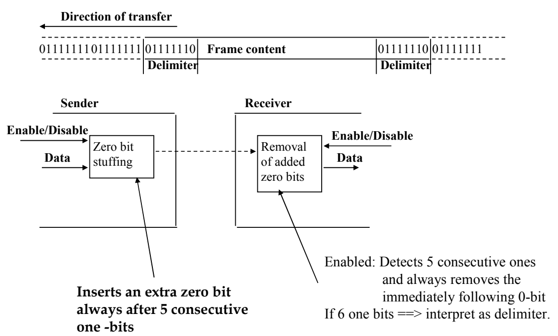
NB: It is also possible that the user buys the NT1 box himself or herself.

Examples of line codes



No direct current component in these codes. This is desirable.

HDLC - transfers frames, delimited by 0111 1110 delimiters.



ISDN D-channel layer 2 = LAPD -protocol is based on this HDLC principle.
HDLC - High level Data Link Control

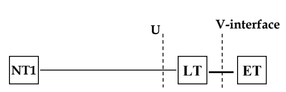
HDLC Sender

```
Do forever
    count=0
    Do While Enable
    if databit=1
        count=count+1
        if count = 5
            send 0
            set count = 0
        fi
    else
        set count = 0
    fi
    send databit
    End do while enable
    If disable
        send 01111110
    fi
End
```

HDLC Receiver

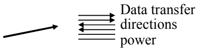
```
Do forever
    When 01111110 received
    count=0
    set Enable
    Do While Enable
    if databit=1
        count=count+1
        if count = 6
            01111110 received !
            set disable
        else
            pass databit (to) onwards
        fi
    else if databit = 0
        if count = 5
            remove 0
        else
            pass databit (to) onwards
        fi
        set count = 0
    fi
End while enable
if disable
    remove tail 011111 from onwards
    pass frame from onwards to the upper
        layer
fi
End do forever
```


(U ja V) -interfaces



- **Network Termination NT1 is connected to the exchange Line Termination using the U-interface.**
  - **Data transfer takes place on a twisted pair copper cable (BRI), the bit rate is 160 kbit/s bi-directionally (full duplex).**
  - **In Finland (originally in US) multi level code 2B1Q is used (-> baud rate is 80 kbaud).**
  - **Bi-directional full duplex transfer is based on echo cancellation: both parties send at the same time, receiver deducts what it has just sent, gets what the far end has sent!**
- **On the V-interface a number of specifications may be used.**
  - **V1-interface applies for the Basic rate interface.**
  - **V3-interface is meant for PBXs with a capacity of 30B+D_{64} channels.**

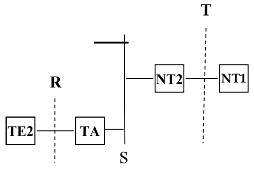
(S, T) -Interfaces



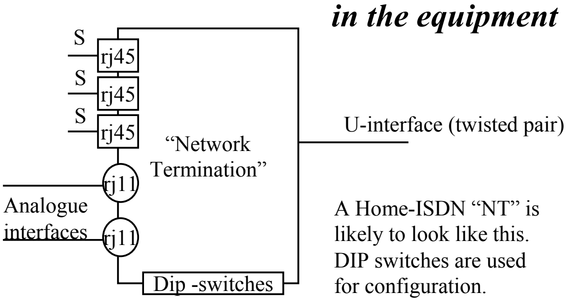
- **S-interface is meant for terminals.**
  - **The interface is a bus by structure**
  - **8 ISDN terminals can be connected.**
  - **Transfer in both directions uses 4 wires.**
- **T-interface is meant for PBXs**
  - **Resides between network termination NT1 and an ISDN-PBX (= NT2).**
  - **T0-interface = S0-interface and is used in PBXs that can serve only BRI connected users.**
- **T2-interface is meant for corporate PBXs**
  - **Transfer rate is 2048 kbit/s.**
  - **T2-interface has 32 channels with 64 kbit/s. Of those 30 are normal B-channels, one is the D-channel and one is used for synchronization. I.e. the structure is like a PCM.**
  - **Other equipment such as Voice Mails and Voice response systems use ISDN primary rate as well.**

R -interface

- ✓ R-interface separates the Terminal Adapter and a non-ISDN device from each other. It follows some existing specification understood by the non-ISDN device (e.g. V.24, V.35 or X.21 - protocol specification).



In practice logical functions are grouped in the equipment



A Home-ISDN "NT" is likely to look like this. DIP switches are used for configuration.

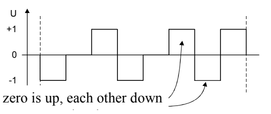
Here physical "NT" = NT1+TA(for analogue phones)

NB: Note the use of rj45 connectors instead of rj11 in Ethernet and also in telephones!

Communication between NT and a Terminal

- ✓ AMI -line code is used between a Terminal and the NT (AMI, Alternate Mark Inversion).

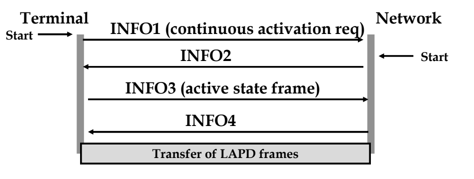
Sent bit	0	1	0	0	1	0	0	0
AMI signal	-1	0	1	-1	0	1	-1	1



- ✓ When there is no traffic over an ISDN interface, terminals are deactivated. A continuous INFO 0 signal is on the interface.

Activation of the basic rate interface facilitates power saving while subscriber is not using the line

- ✓ Terminal starts activation by sending a continuous activation request: INFO 1. When the network detects the request, it starts sending synchronization frames INFO 2.
- ✓ When the network initiates activation, it starts sending INFO 2 directly.
- ✓ When the terminal detects a synchronization frame, it stops sending the activation request signal and starts sending active state frames INFO 3. When the local exchange has received active state frames, it moves to INFO 4 state. The physical layer is now active and ready for information transfer.



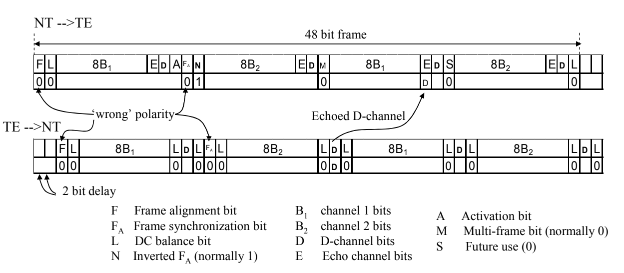
This activation procedure facilitates power saving: the exchange does not need use power for inactive interfaces. After the activation layer 1 is working and the signaling itself can start on layers 2 and 3.

Activation and call setup

- ✓ Prior to activation there is no power in the interface and no bits are transferred.
- ✓ After activation there is a continuous stream of bits on the interface in both directions
  - › This means that layer 1 is ready for requests from layer 2
  - › In 2B+D interface also a limited power feed from the local exchange is available for emergency calling under a mains power failure.
- ✓ Layer 2 works on the signaling channel only and transfers frames
- ✓ Inside layer 2 frames, in their payload, Q.931 signaling messages are sent.
- ✓ When layer 2 is ready (it has addresses for the endpoints etc), layer 3 can start a call setup procedure

Frame structure on the S-interface

- ✓ 48 bit frames 4000 times per second are used between a terminal and the NT1.
- ✓ The resulting bit rate is 192kbit/s



Frame synchronization on S-interface

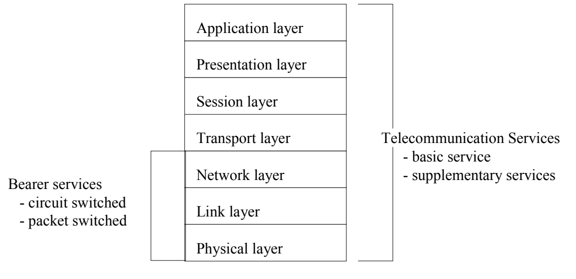
- ✓ Frame synchronization is achieved by sending violation bits in the AMI code.
- ✓ The first (F) and the 14th bit (FA) equal to zero with a wrong polarity I.e. the same as the previous zero. To balance this for the sake of zero average voltage, the wrong zero is followed by DC balance bit (L).

NB: On S-interface the AMI code is inverted, I.e. logical zero is sent as a pulse with alternating polarity and a logical 1 is sent as zero voltage.

Overhead bits in the frame carry D-ch echo and control power consumption

- ✓ A Terminal can see that the NT has received its D-channel bits based on E(echo)-bits. NT copies a received D-bit to the next E-bit.
  - › If two terminals are signaling at the same time on the S-interface, wrong E-bits tell that there was a collision and that the terminal should wait for a while prior to sending again.
- ✓ A-bit is used for power control. With A-bit, the network can command the terminals to deactivate themselves and to transfer to a low power mode in which they are only able to become active again either on network request or user action. The activation procedure uses the A bit.

Message based signaling can be functionally split following the OSI 7 layer model



Bearer services are transport services that are seen by the "user"

- ✓ Circuit switched bearer services include:
  - › Speech
  - › 3,1 kHz audio
  - › 7 kHz audio
  - › transparent 64 kbit/s.
- ✓ Packet switched bearer services include:
  - › virtual call and permanent virtual connection,
  - › connectionless packet switched service on the D-channel,
  - › user-to-user signaling information.

Telecommunication Services incorporate all OSI layers

- ✓ A Telecommunication service is a set of functions offered to a user and it is implemented using the capabilities of all OSI layers.
- ✓ Telecommunication services make use of the bearer services.
- ✓ Telecommunication services can by further divided into basic and supplementary services.
- ✓ Supplementary services can be used only in connection with a basic service.
- ✓ The term "feature" is more generic than "supplementary service". In addition to supplementary services it refers to any functional properties of a system. Sales arguments and sales contract often list a lot of features…

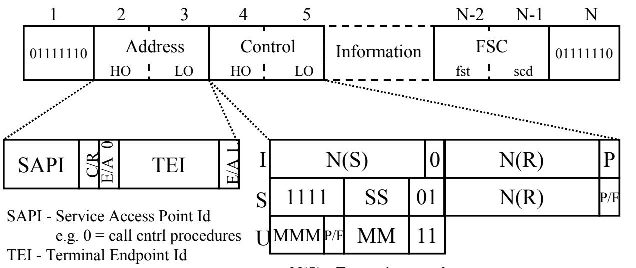
Digital Subscriber Signaling System No 1 (DSS 1)

- ✓ DSS1 is based on a protocol stack that includes three OSI lower layers.
- ✓ DSS1 is fully message based and out-of-band offering the possibility of signaling while the voice channel is open end-to-end.
- ✓ DSS1 messages are sent on the D-channel.
  - › NB: tones: dial-tone, ringing tone, busy tone are sent by exchanges on the audio channel.
- ✓ DSS1 layer 2 follows the HDLC principles and is called the LAPD-protocol (Q.920 - Q.921).
- ✓ DSS1 signaling overview is given in ITU-T Q.930 and detailed procedures are given in Q.931.

Q.920/Q.921 - LAPD

- ✓ **Connectivity over the link between a terminal and the Local exchange**
  - › **Inherits HDLC principles.**
  - › **Corresponds to the OSI layer 2 requirements**
- ✓ **Transfers frames from *many terminals to many layer 3 entities.***
- ✓ **Properties:**
  - › **DLCI - data link connection id identifies the link connections: DLCI = SAPI + TEI. SAPI = Service Access Point Id, TEI = Terminal Endpoint Id - are purely layer 2 concepts. Layer 3 uses CEI - Connection Endpoint Id = (SAPI+CES)**
  - › **Can guarantee frame order due to numbering.**
  - › **Fault management - lighter than MTP in CCS#7.**
  - › **Flow control based on windowing (windowing means that e.g. N=window size messages can be sent before an acknowledgement is required).**

LAPD frame format



SAPI - Service Access Point Id
e.g. 0 = call cntrl procedures
TEI - Terminal Endpoint Id
E/A - Address field extension bit
= 1 = final bit
C/R - Command/Response field bit
I / S/ U – types of frames.

N(S) - Transmitter send sequence nr
N(R) - Transmitter receive seq. Nr
S - Supervisory function bit (e.g. ack frames)
M - Modifier function bit
P - poll when issued as a command,
final bit when issued as response

Q.921 - LAPD

- ✓ **Point-to-point link connections, multi-point connections - broadcast**
- ✓ **Initiation state - TEI values not yet chosen.**
  - › **Before any higher level (Q.931) functions can be performed, each ISDN device must be assigned at least one unique TEI value. These numbers can be preassigned (TEIs 0-63), or dynamically assigned (TEIs 64-126). Most TEI assignment is done dynamically, using the TEI management protocol.**
- ✓ **Unnumbered Information = UI -frames are not acknowledged**
  - › **also broadcast (e.g. SETUP to B subscriber)**
  - › **faults recovery is left for the upper layers.**
- ✓ **Acknowledged mode - I - numbered frames**
  - › **fault recovery and flow control procedures supported on layer 2**

NB: LAPD is between TE and the local exchange. Its sole purpose is to carry frames between the two taking into account that many terminals can be connected to an S-interface.

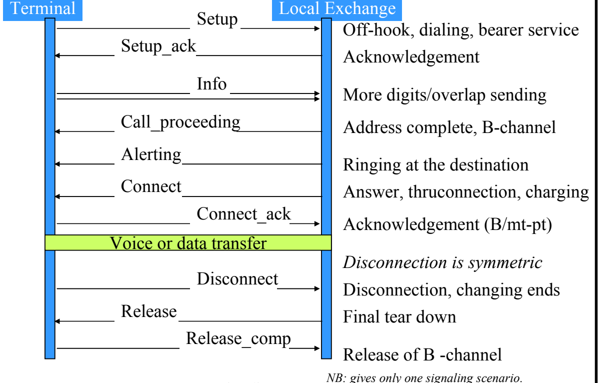
DSS1 - Q.931 - signaling

- ✓ **Corresponds to layer 3 - network layer:**
  - › **understands end-to-end addresses: E.164 telephone numbers**
- ✓ **Can set up, control and release circuit switched calls.**
- ✓ **Supports also packet switched on-demand connections.**
- ✓ **Call identification is based on the *call reference* - and has nothing to do with e.g. the identity of the B-channel in use!**
- ✓ **Supports the functional and the stimulus (keypad) modes of signaling.**
- ✓ **User-to-user information transfer in signaling messages is also supported (charging is an issue).**

Functional and stimulus -modes

- ✓ **Functional**
  - › **Information is encoded in service specific information elements.**
  - › **As a result, signaling becomes *service dependent.* A new service requires new programs both in the CPE and the exchange**
  - › **Can be OK, if CPE = PBX**
  - › **For phones would really require a JAVA -like automatic software download function. There is no such thing in ISDN!**
- ✓ **Stimulus -mode:**
  - › **phone button pushes are carried in signaling as such (a field in a message tells which button was pushed)**
  - › **Interpretation is the responsibility of the exchange**
  - › **A new service requires new programs only in the network**
  - › **The phone may have programmable soft keys to hide dialing sequences**

Q.931 -signaling call setup procedure



NB: gives only one signaling scenario.

DSS1 (Q.931) call signaling

- ✓ DSS 1/Q.931 signaling is symmetric. This means that originating call signaling and terminating signaling use the same messages, just the direction is (ex –terminal) is reversed. Symmetry applies to both call setup and release.
- ✓ The SETUP message contains at least the bearer service (= is this a data call or audio call). In practice it is best to include as many digits in the SETUP message as there are in the shortest directory number in the local dialling plan (4 in the Finnish ISDN network e.g. 4511 for TKK), so the originating exchange can attempt routing immediately having received the SETUP message.
- ✓ SETUP_ACK acknowledges the reception of SETUP. It is more useful on the terminating side – tells that the kind of device that can support the bearer and telecommunication service requested has been found (e.g. a G4 fax machine).
- ✓ INFO messages support overlap sending – the result is that the routeing can be done through the network with the minimal number of digits (in Finland usually 3). NB1: In case there is no number portability (NP), the last digits (in Finland 4…8) are needed only at the terminating exchange, the rest of the exchanges do not even look at the last digits. If NP is allowed in the number block, all digits need to received prior to routeing decision at the originating exchange. NB2: There is no such thing as number complete indication from the terminal, instead the network deduces when all digits have been received by executing number analysis after each digit or at times instructed by former routing information retrieved based on earlier digits received for the call.
- ✓ CALL_PROCEEDING – tells that no more digits will be needed even at the terminating exchange, it also tells that at least the network is not busy (the B-subs can still be busy)
- ✓ ALERTING tells that the phone at B-subscriber (called party) is ringing. The D-channel message is accompanied by a ringing tone from the terminating exchange to the caller on the audio channel. This tells that the audio challel is clear at least in the backward direction from the terminating exchange to the caller.
- ✓ CONNECT message starts charging and all exchanges throughconnect the B-channel in both directions.
- ✓ CONNECT ACK is really necessary at the terminating side: tells which of max of 8 devices that can be connected to the S-interface the call has been awarded (imagine two or more phones being picked up almost at the same time each sending a CONNECT).
- ✓ DISCONNECT – charging stops and the B-ch connection is torn down.
- ✓ RELEASE tells that DISCONNECT was received. RELEASE COMPLETE confirms the reception of RELEASE and tells that after this the B-ch can be used for another call on the subscriber interface. For call tear down in exceptional cases even RELEASE and RELEASE COMPLETE are enough.

Some details

- ✓ Overlap sending means that DSS1 supports the traditional dialing procedure known from analogue phones.
  - › The orginating exchange can start number analysis and routing when it has received enough digits. Usually it will make the first attempt of routing when it has received the amount of digits in the shortest directory number in the dialling plan.
  - › alternatively the GSM style dialing (select all digits first and then push the call button) can be used as well. In that case all digits are contained in the one and only SETUP message.
- ✓ Tones
  - › Ringing phase: alerting message is accompanied by alterting tone on the audio channel from terminating exchange to A-subscriber.
  - › If call setup fails, busy tone can be sent in the backward direction from any exchange on the call path.
- ✓ Windowing (in LAPD) means that the sender can send a max of say N messages without getting an acknowledgement. Having sent N messages it must wait for an acknowledgement.
  - › one acknowledgement can typically acknowledge all messages to a point in the stream of messages.
- ✓ NB: DSS 1 does not follow the client server model: it is based on the concept of *communicating finate state machines*.

Addressing of users in DSS1

- ✓ Called Party and Calling Party Number = E.164 telephone numbers
  - › Directory numbers
  - › Also C-party number in case of Fall Forwarding
- ✓ The max 8 devices connected to one S-interface may have one or different telephone numbers
  - › If only one number is used Low level Compatibility Information and High Level Compatibility Information in the SETUP message can make a distinction between a telephone, a fax machine, a computer etc.
  - › When the devices with the same telephone number all receive SETUP at the same time from the network, they look at the compatibility info and respond based on match.
  - › Connect shows which of the devices the user used to answer.
  - › Connect_ack shows which of the possible Connects won the race and was received first by the network.
- ✓ Subaddress allows extending E.164 addressing

Use Case: Connecting PABXs to a Public Network

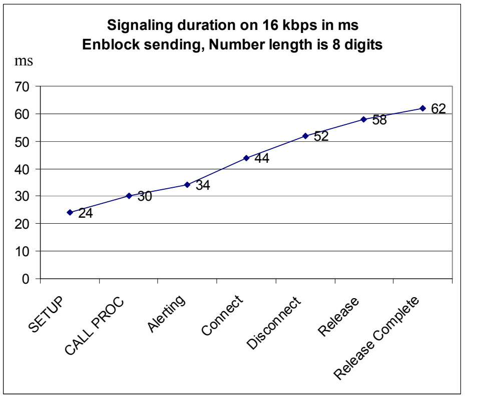
- ✓ Numbering and DDI:
  - › A PABX can have a private numbering plan (i.e. a plan that is not visible in the public network). In that case DDI – direct dialling in is not possible, instead all incoming calls are connected through the switchboard of the PABX. This sets mininal requirements for signaling between PABX and the public network – even analogue subscriber signaling will do the job.
  - › If PABX numbering plan is a subset of the public network plan, DDI becomes possible and when an incoming call is delivered, some digits need to be sent from the public net to the PABX. Analogue subscriber signaling is not enough any more. E.g. digital R2 can be used instead.
- ✓ Typical signaling systems PABX to ISDN exchange are primary rate ISDN (Q.931), DASS (UK). In these cases all business services to PABX extensions are provided by the PABX.
- ✓ Private networking:
  - › PABXs can be connected to private PABX networks (multi-site companies). Such networks can be sometimes extended with public network extensions – in that case the hosting public exchange becomes part of a private network as well. Some PABX services require PABX to PABX signaling to provide them in a private PABX network. In that case the public switch or switches need to support e.g. Q.SIG – a private network signaling system, or DPNSS – a UK private network signaling system.
  - › NB: many private network signaling systems are proprietary (vendor specific) although may be based on DSS1

A Signaling dump from an access line

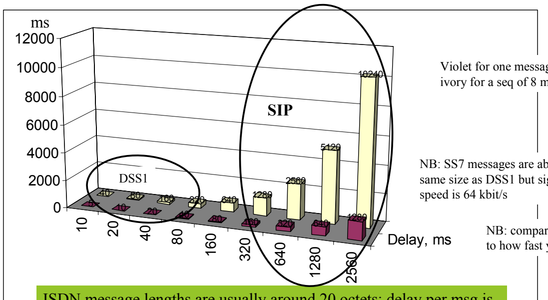
SETUP	**00990000**0801130504038090A31801836C020080700380333327D0291817E0104
CALL PROC	**02990002**0801930218018A
Alerting	**02990202**08019301
Connect	**02990402**080193072905690107014247C038090A3
Disconnect	**02990602**0801934508028190 1E028188
Release	**00990208**0801134D08028090
Release Complete	**02990804**0801935A

- ✓ Messages are presented in hexa code
  - › Hexa code is 0, 1, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F, where e.g A =1010 in binary
- ✓ Bold characters are Layer 2 bits
- ✓ Normal characters are Layers 3 (Q.931) bits
- ✓ The setup message carries just two digits

ISDN signaling duration on 16kbps (L2+L3)



Message duration on 16kbit/s channel per message and per 8 message sequence



Violet for one message
ivory for a seq of 8 msgs

NB: SS7 messages are about the same size as DSS1 but signaling speed is 64 kbit/s

NB: compare these to how fast you dial!

ISDN message lengths are usually around 20 octets: delay per msg is around 10 ms and for the whole sequence less than 100 milliseconds.

message duration in ms = message length in bits including all layers/signaling channel speed in kbit/s

Signaling efficiency

- ✓ **Delay per message, delay for the call setup flow, delay for a call flow including setup and release.**
  - › **Can be calculated directly from specification**
- ✓ **Post dialling delay: from the moment last digit is pushed by the caller till the caller hears ringing tone.**
  - › **Modeling requires the knowledge of network structure!**
  - › **Close to user perception**
- ✓ **Number of bits in a call flow**
  - › **All bits in all calls create signaling traffic – the amount of signaling traffic should be significantly less than the signaling channel capacity**
  - › **Users pay for the service: businesswise signaling is pure overhead – it follows that the following should apply**
    - **Amount of signaling bits in a call flow << amount of bits in the service**
  - › **Amount of bits in the service is e.g. amount of voice bits in a call**
  - › **If calling is a flat rate service, amount of signaling bits is not that important**

Message transfer delay in previous slides

$$\text{Msg transfer delay} = \frac{\text{Message length in bits}}{\text{Signaling channel speed}}$$

Naturally the same formula applies to message flows.

Assume: message length = 1 000 bits and 16 kbit/s signaling channel

$$\text{Msg transfer delay} = \frac{1000 \ \text{bits}}{16\,000 \ \text{Bits/s}} \approx 60 \text{ ms}$$

ISDN Summary

- ✓ **Signaling and voice channel are both physically and logically separated (out-of-band, common channel + call reference).**
- ✓ **Any signaling info needed for services is supported or can be added.**
- ✓ **Q.931 signaling is service dependent, contains really information that is relevant on OSI-layers 3 - 7! New services require new programs in CPEs in case of functional mode. There is no mechanism for automatic software download to CPEs.**
- ✓ **Multi-point structure complicated the implementation significantly.**
- ✓ **Major consumer value added is in 2 x 64kbit/s bit rate. ISDN adoption is determined by home Internet use.**
- ✓ **ISDN specified a digital PBX access signaling for the first time. This has been widely adopted! ISDN signaling has been reused in many new signaling applications (V5, private PBX networks, IP-Telephony, conferencing, GSM etc.) So, ISDN (DSS1) is the mother of many modern signaling systems.**

DSS 1 strenths

- ✓ **Cmp to analogue signaling:**
  - › **out of band: signaling can continue during a call. This gives flexibility in service implementation**
  - › **fast: signal delays in the order of tens of ms per signal. Low post dialling delay if network signaling is fast also (such as SS7)**
  - › **many symbols: can support potentially a very wide range of services**
  - › **reliable: HDLC on layer 2 with msg numbers and checksums**
  - › **physically and logically separate from voice channel: flexibility for new services**
  - › **"native signaling" for digital exchanges and terminals: no AD conversions. Information is sent pretty much in formats that can directly be processed by a program controlled device.**
- ✓ **Historically or in hindsight**
  - › **provided digital signaling for PABXs, allows e.g direct dialling in (which could also be provided by R2 but not all analogue signaling systems**
  - › **became mother of many PABX signaling systems (QSIG and proprietary variants), mother of GSM signaling, etc. (also mother of V5 access signaling, mother of B-ISDN signaling but these are not successes)**

DSS 1 weaknesses (all in hindsight)

- ✓ **Technical**
  - › **functional mode requires new software both in terminals and exchanges when new services are introduced. However, DSS1 does not have software download from the network. Stimulus mode requires new software for new services at least for exchanges. We say that such signaling is service dependent.**
  - › **A 2B+D line card could serve approximately half as many subscriber lines as an analogue subscriber line interface card. Therefore ISDN subscirber access was more expensive than analogue access. NB: about 70% of local exchange cost is in line cards.**
- ✓ **Value**
  - › **ISDN (2B+D) provides limited added value for subscribers, therefore penetration started to grow only when Internet and www became popular and higher bandwidth than modems could provide were needed.**
  - › **Closed market solution: does not provide innovation opportunities for any users like the Internet does. You are stuck with the services provided by your operator. Operators are stuck with the services provided by a small number of vendors. ISDN sw is always provided by equipment manufacturers for exchanges and for terminals. Neither does ISDN open particular opportunities for competitive operators.**