Chapter 12:
Voice over ATM and MPLS

- TOPICS
  - Background
  - Voice over ATM specifications
  - Voice over MPLS

Why voice over packet?

- *National and international operators*
  - It permits them to maintain a single group of engineers and managers to run a single network.
  - Single network management.
- *Alternate carriers or value added network suppliers (CLECs)*
  - Cost and limited availability of bandwidth necessitates packet switching.
  - Within the local loop, they can provide voice and data services over cable modem, ADSL and APON.

- *Cellular operators*
  - They have to interconnect their cell sites and *Message Switching Centers* (MSC).
- *Private (enterprise) networks*
  - A significant portion of the traffic is voice. Since they buy bandwidth at commercial rates, integrating voice and data is a cost-effective solution.

ATM Forum specifications

- *ATM Trunking for Voice* is a group of specifications that deals with the transport of voice between two telephone networks.
- We describe the following two specifications:
  - *Circuit Emulation Services* (CES),
  - *ATM Trunking Using AAL 2 for Narrowband Services.*

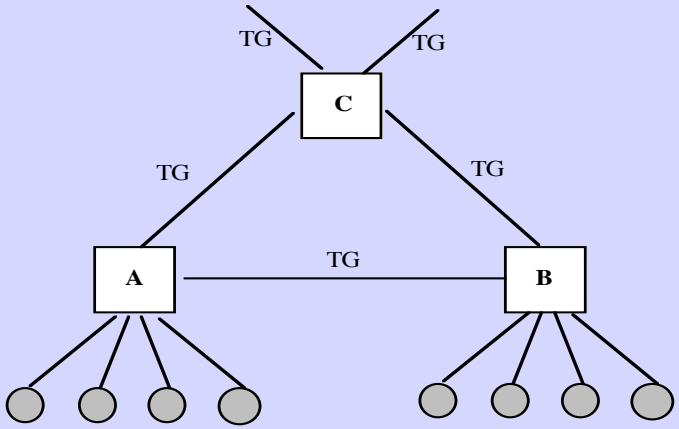
MPLS and Frame Relay Alliance specifications

- Two specifications have been defined based on ATM's AAL 1 and AAL 2 protocols.
  - Circuit emulation services over MPLS using AAL 1.
  - Voice over MPLS using AAL 2.

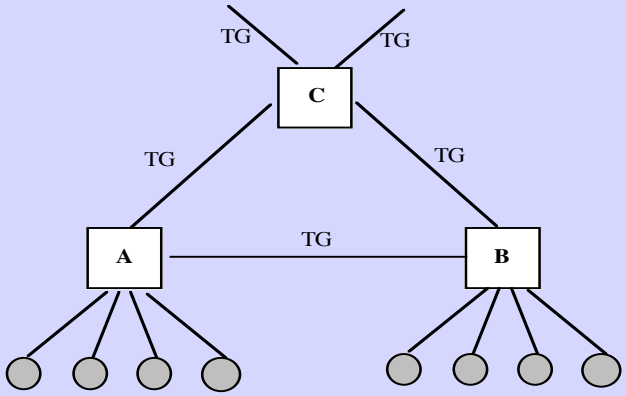
Background

- Topics
  - Some basic concepts
  - Channel associated signaling (CAS)
  - Signaling System No. 7 (SS7)
  - Narrowband ISDN (N-ISDN)
  - Digital subscriber signaling system No. 1 (DSS1)

Some basic concepts



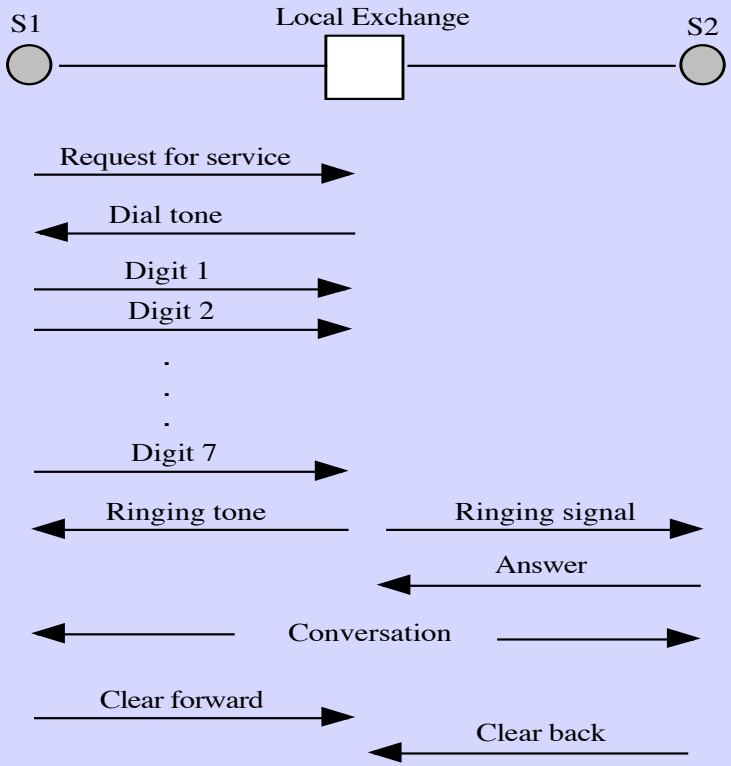
- A and B are known as *local exchanges*, or *central offices*, or *end offices*.
- C is known as an *intermediate* or *tandem*, or *toll*, or *transit exchange*.
- *Trunk:* is a circuit between two exchanges. It is nothing else but a channel associated with a time slot in a T1/E1 link or in a SONET/SDH link, which carries a single voice call.
- A group of trunks is known as a *trunk group* (TG).



- A *subscriber or customer, or user* is connected to its local exchange via a *subscriber line*, which is commonly referred to also as the *local loop*.
- *Private Branch Exchange (BPX):* An exchange owned by an organization (University, business, government, agency). Employees in the organization can call each other and also place and receive calls tø from outside the organization.

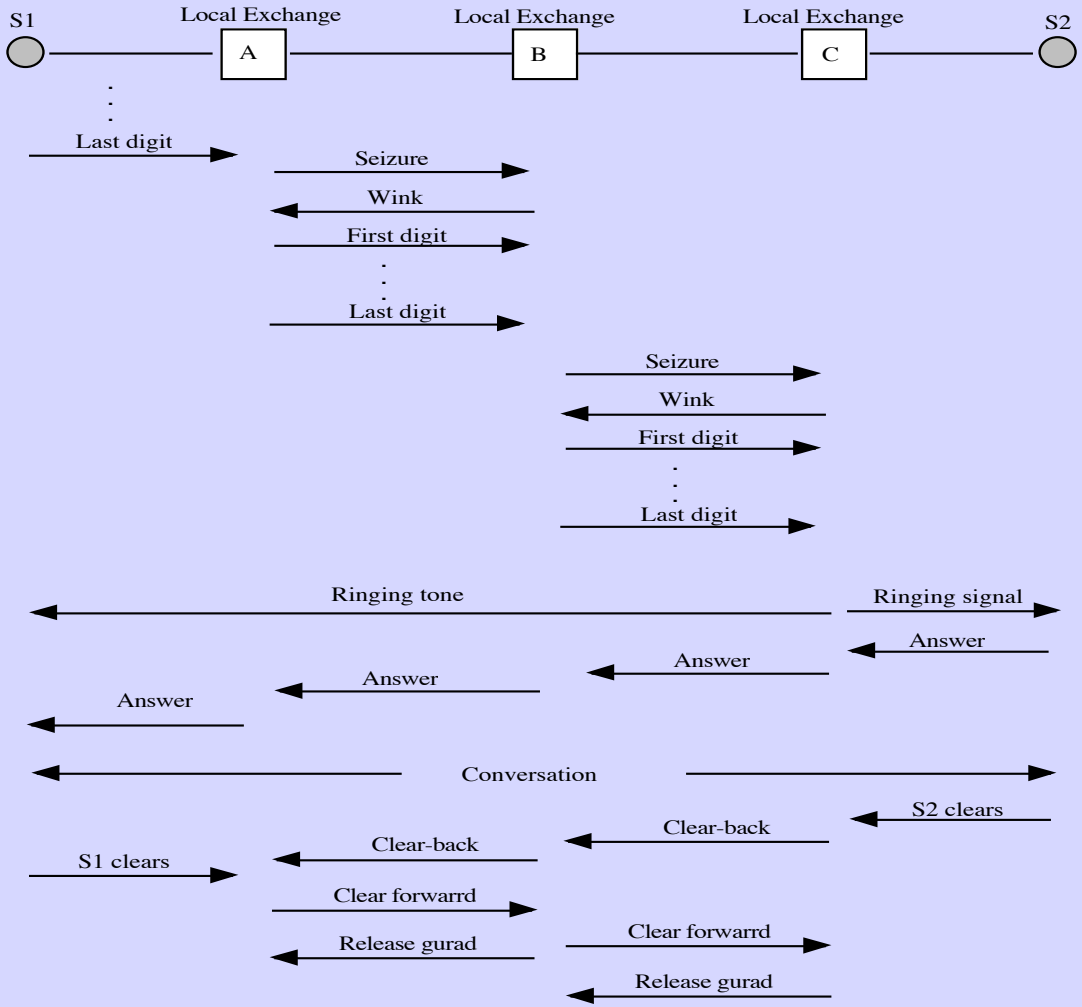
Basic signaling

- Early telephones used *dial-pulse* to generate the digits.
- The dial-pulse system was replaced by the *dual-tone multi-frequency* (DTMF) system. When a subscriber presses a key on the keypad, an oscillator inside the telephone generates two simultaneous tones.
- The DTMF frequency combinations are distinct from naturally occurring sounds, voice and voiceband data


S1
Local Exchange
S2
Request for service
Dial tone
Digit 1
Digit 2
Digit 7
Ringing tone
Ringing signal
Answer
Conversation
Clear forward
Clear back

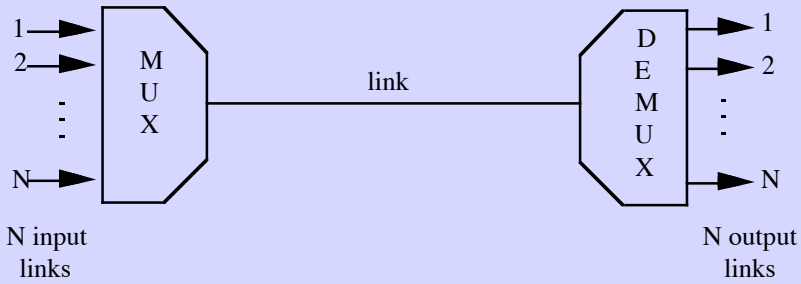
Channel-associated signaling (CAS)

- It was the only signaling system used until the late 1970s, when the *Common Channel Signaling (CCS)* was developed.
- CAS is still being used, but it is gradually been replaced by CCS


S1
Local Exchange
A
Local Exchange
B
Local Exchange
C
S2
Last digit
Seizure
Wink
First digit
Last digit
Seizure
Wink
First digit
Last digit
Ringing tone
Ringing signal
Answer
Answer
Answer
Conversation
S2 clears
Clear-back
Clear-back
S1 clears
Clear forwarrd
Release gurad
Clear forwarrd
Release gurad

- In addition to the signals: seizure, wink, answer, clear-forward, clear-back, and release guard, and the dialed digits which are transported using the DTMF scheme, a number of supervisory bits are also used to indicate supervisory line states such as on-hook, off-hook, idle, and ringing.
- These signals are transferred using the ABCD signaling bits

The ABCD bits



TDM multiplexing

The DS-1 signal

- The 8th bit of every sixth time slot in a voice channel is "robbed" and it is used to transmit signaling information.
- The robbed bit can be used in the following three schemes:
  - a) *AB signaling*,
  - b) *ABCD signaling*,
  - c) *network control signaling*.

AB signaling

- In AB signaling, the 8^{th} bit of each time slot in every sixth frame is used to transmit the A bit and the B bit alternatively.
- That is, the A bit is transmitted on the 6^{th} frame, 18^{th} frame, 30^{th} frame and so on.
- The B bit is transmitted on the 12^{th} frame, 24^{th} frame, 36^{th} frame, and so on.

The AB signaling scheme

Frame	Information	Frame	Information
1	8-bit voice	13	8-bit voice
2	8-bit voice	14	8-bit voice
3	8-bit voice	15	8-bit voice
4	8-bit voice	16	8-bit voice
5	8-bit voice	17	8-bit voice
6	7-bit voice, A	18	7-bit voice, A
7	8-bit voice	19	8-bit voice
8	8-bit voice	20	8-bit voice
9	8-bit voice	21	8-bit voice
10	8-bit voice	22	8-bit voice
11	8-bit voice	23	8-bit voice
12	7-bit voice, B	24	7-bit voice, B

ABCD signaling

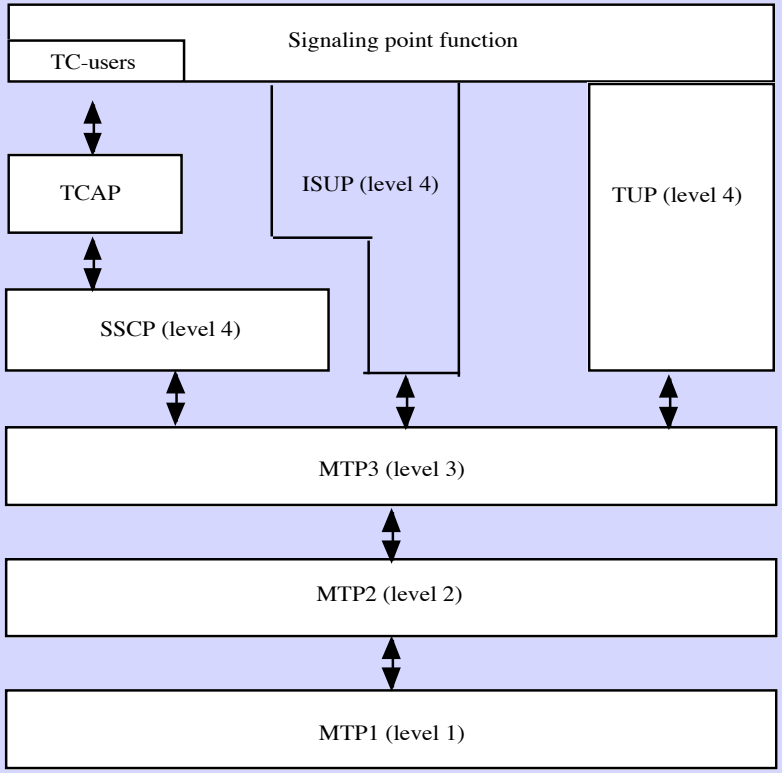
Frame	Transmitted bit
6	A
12	B
18	C
24	D
30	A
36	B
42	C
48	D

- The 8th bit of each sixth time slot is robbed as in the A,B scheme.
- The robbed bit is used to transmit the A, B, C, and D bits alternatively.

Signaling System No. 7 (SS7)

- In CAS, the signaling information for a trunk is carried in the trunk itself using the ABCD bits and the DTMF scheme.
- In *Common-Channel Signaling* (CCS), all signaling information, including dialed digits, is carried in messages over a separate packet-switched network.

The SS7 stack



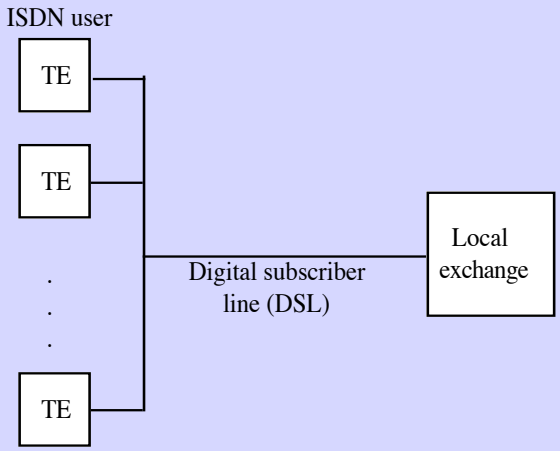
Narrowband ISDN

- The *Integrated Service Data Network* (ISDN) was a concept of a future network put forward in the 1980s.

This future network will be capable of providing a wide range of services for voice and non-voice applications, and all these services will be provided by one network, rather than different networks.

- The first generation of ISDN is referred to as the *Narrowband ISDN* (N-ISDN). ISDN users can communicate with each other in circuit-switched mode and packet-switched mode.
- The second generation of ISDN that provides for very high speeds was referred to as the *Broadband ISDN* (B-ISDN). The *Asynchronous Transfer Mode* (ATM), is a packet-switched network architecture used for B-ISDN.

An ISDN user



- Multiple 64 Kbps digital *Terminal Equipment* (TE) of several types can be used, such as,
  - *digital (PCM) telephone,*
  - *high-sped facsimile terminal*
  - *computer modem.*
- The TEs are connected to the local exchange via a *Digital Subscriber Line* (DSL). The DSL is a 2-wire or 4-wire line that allows simultaneous transmissions in both directions.

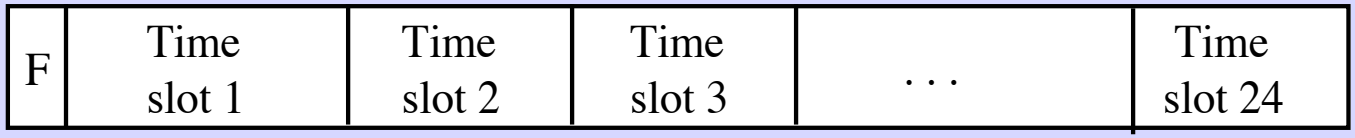
Basic rate: 2B + D

- The transmission rate in one direction is 144 Kbps.
- The 144 Kbps bit stream is divided into *two 64 Kbps B-channels and one 16-Kbps D-channel.*
  - B-channels: used for circuit-mode communications
  - D-channel is used for signaling between the user and the local exchange - *Digital Subscriber Signaling System No 1* (DSS1) and for low speed packet-switching.

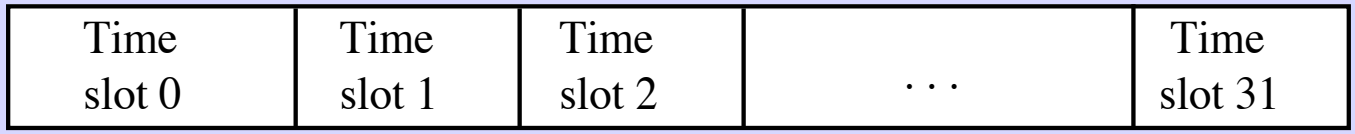
Primary rate: 23B+D

- Defined for users with greater requirements for bandwidth, such an organization with a digital PBX or a local network.
- In USA, Canada, and Japan the primary rate is 1.544 Mbps.
- In Europe, the primary rate is 2.048.

Primary rate frame structures



a) Frame structure for the 1.544 Mbps interfcae



a) Frame structure for the 2.048 Mbps interfcae

Digital subscriber signaling system No. 1 (DSS1)

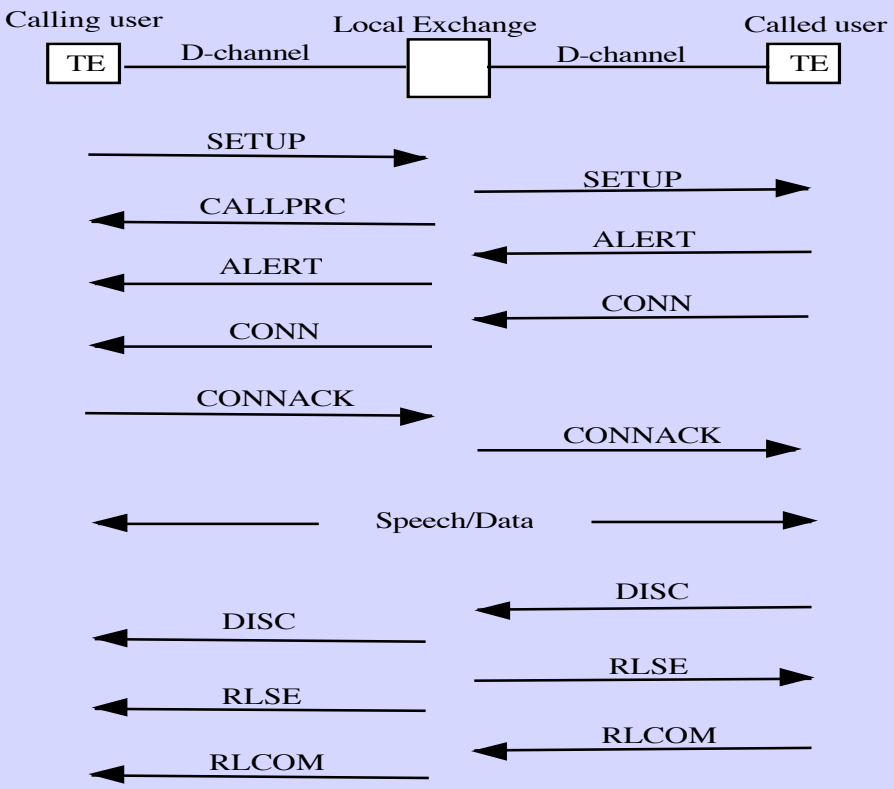
- The *Digital Subscriber Signaling System No.1* (DSS1) is used for signaling between a N-ISDN user and its local exchange.
- DSS1 is message oriented and many of DSS1 concepts are similar to SS7.
- DSS1 messages are transported over the D-channel.

- DSS1 is divided into the *data link layer* and the *network layer*.
- The data link layer is also known the LAP-D link access protocol, and it is concerned with the reliable transfer of frames between the terminal equipment (TE) and its local exchange.
- The network layer protocol was defined in ITU-T recommendation Q.931.
- ATM's Q. 2931 is based on Q.931, and it also referred to as the *Digital Subscriber Signaling System No.2* (DSS2).

Q.931 messages

- *set-up (SETUP),*
- *set-up acknowledgment (SETACK),*
- *call proceeding (CALPRC),*
- *progress message (PROG),*
- *alerting message (ALERT),*
- *connect (CONN),*
- *connect acknowledgment (CONACK),*
- *disconnect (DISC),*
- *release (RLSE),*
- *release complete (RLCOM), and*
- *information (INFO)*

An example of DSS1 signaling



Voice over ATM

ATM specifications:

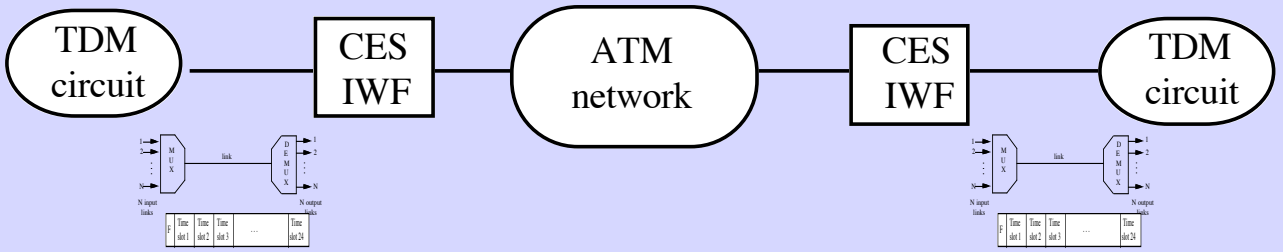
1. ATM trunking using AAL 1 for narrowband services
2. Circuit emulation services (CES)
3. Dynamic bandwidth circuit emulation services
4. ATM trunking using AAL 2 for narrowband services
5. Voice and Telephony over ATM to the Desktop
6. Loop Emulation Service (LES) using AAL 2.

Circuit emulation services (CES)

- CES emulates a point-to-point TDM circuit.
- CES is used to connect TDM interfaces such as such as T1, T3, E1, E3, and J2, over an ATM network.
- CES is based on AAL 1 and it uses the CBR service category in order to guarantee the end-to-end delay. Both the unstructured and structured data transfer protocols of AAL 1 CS sublayer.

- The following services have been defined:
  - *Structured DS1/E1 Nx64 Kbps (fractional DS1/E1)*
  - *Unstructured DS1/E1 (1.544 Mbps/2.048 Mbps)*
  - *Unstructured DS3/E3 (44.736 Mbps/34.368 Mbps)*
  - *Structured J2 Nx64 Kbps (fractional J2)*
  - *Unstructured J2 (6.312 Mbps)*

The reference model



- The two CES IWFs are connected by an ATM connection using the CBR service category.
- Each CES IWF is connected to a TDM circuit such as T1, T3, E1, E3, and J2.
- The two IWFs extend transparently the TDM circuit across an ATM network

Structured DS1/E1/J2 Nx64 Kbps service

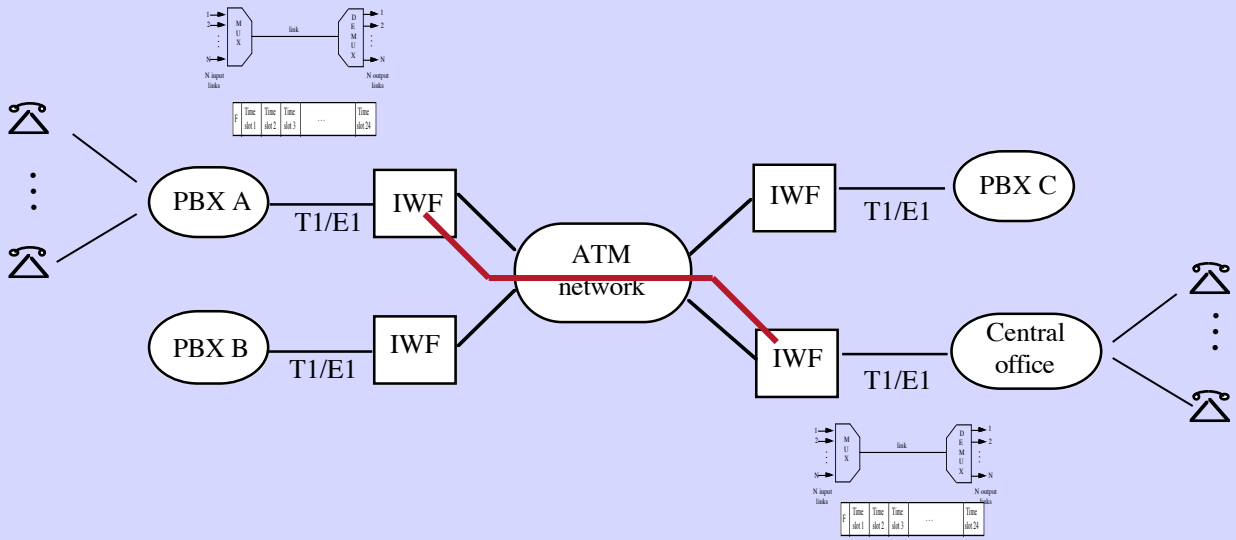
- This service is intended to emulate point-to-point fractional DS1, E1 and J2 circuits, where N takes the values, $1 \leq N \leq 24$ for DS1, $1 \leq N \leq 31$ for E1, and $1 \leq N \leq 96$ for J2.
- The AAL1 CS structured data transfer protocol is used

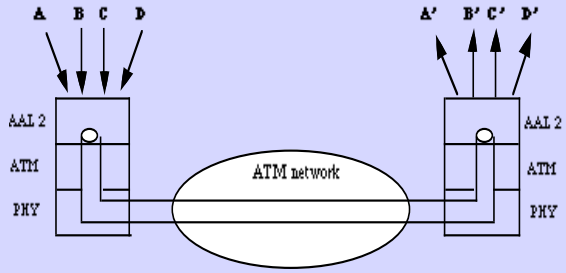
DS1/E1/J2 unstructured service

- The unstructured service emulates a point-to-point DS1/E1/J2 circuit across an ATM network.
- The incoming bits from the DS1 circuit are simply placed sequentially into the payload of the AAL 1 without regard to framing using the unstructured data transfer protocol

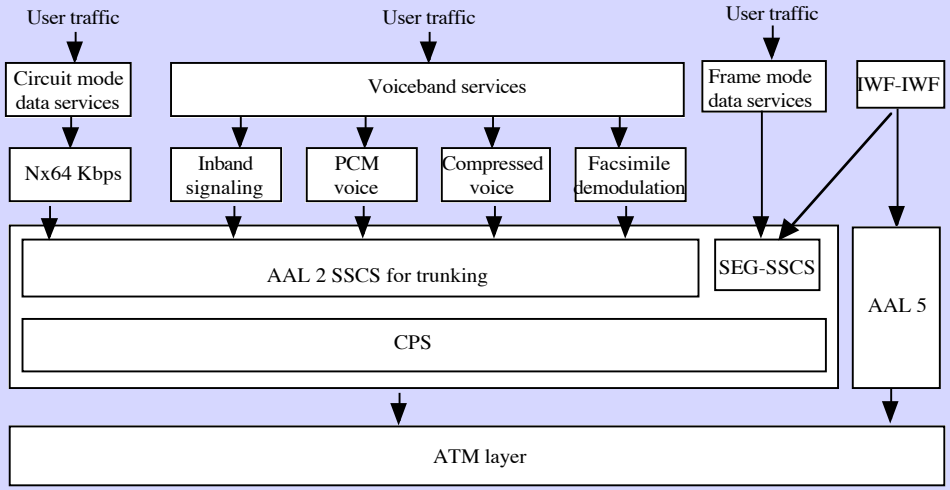
ATM trunking using AAL 2 for narrowband services.

- This specification was designed so that it can be used to interconnect two distant public or private telephone networks over an ATM network.


PBX A
PBX B
T1/E1
T1/E1
IWF
IWF
ATM
network
IWF
IWF
T1/E1
T1/E1
PBX C
Central
office

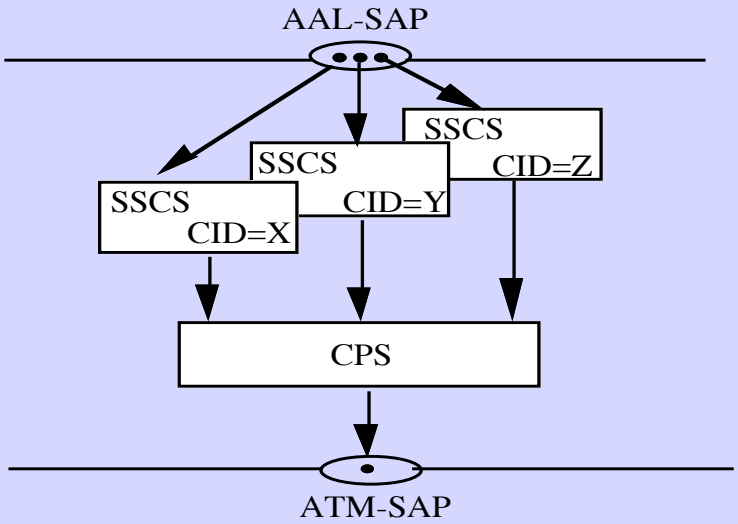

A
B
C
D
A'
B'
C'
D'
AAL 2
ATM
PHY
ATM network
AAL 2
ATM
PHY

The protocol stack of an IWF

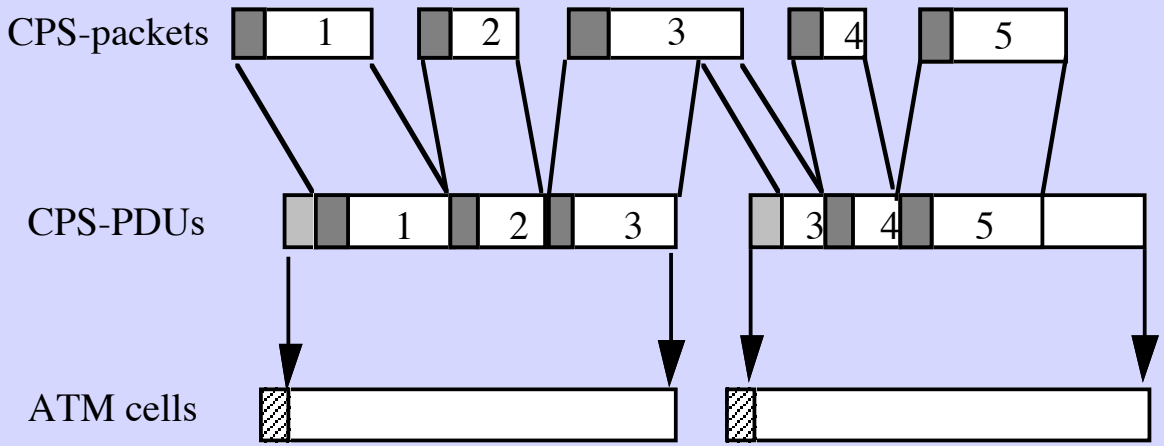


Functional model of AAL 2 (sender side)

- Each SSCS stream is associated with a CID



Packing CPS-packets into CPS-PDUs



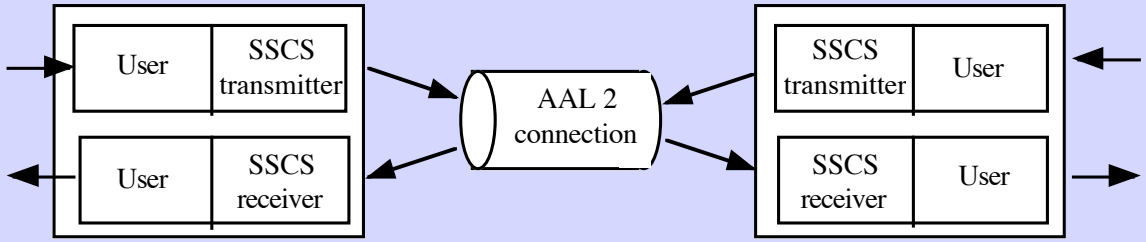
- A specialized SSCS has been developed to support “ATM trunking using AAL 2 for narrowband services”.
  - *AAL SSCSC for trunking*
  - *segmentation and reassembly service specific convergence sublayer* (SEG-SSCS).

AAL 2 SSCSC for trunking

The purpose of the AAL 2 service specific convergence sublayer (SSCS) for trunking, is to convey

- telephone voice calls,
- voiceband data, such as facsimile and data transmitted over a modem, and
- fractional T1/E1 circuit-mode type of data.

The reference model of the AAL 2 SSCS for trunking



- On either side of the connection there is a transmitting and a receiving SSCS.
- For each transmitting or receiving SSCS there is a *User*, a signal processing device that passes to and receives information from the SSCS.

Functions of *User*

The following are some of the functions provided by the *User*:

a) Audio encoding/decoding

b) Detection and extraction of DTFM and CAS bits

c) Facsimile

d) Circuit mode data (fractional T1/E1)

e) Data frames

Audio encoding/decoding

- At the transmitter's side, it encodes audio samples using one of several audio algorithms. The transmitting User also detects silence periods and sends silence insertion descriptors.
- At the receiver's side, it decodes audio bits into a sequence of audio samples, including comfort noise generation as directed by silence insertion descriptors.

Encoding algorithms

- Various encoding algorithms can be used.
- Each algorithm creates encodings which are grouped together into a packet referred to as the *Encoding Data Unit* (EDU).
- Bigger packets can be formed by concatenating several EDUs.
- The destination User, is responsible for decoding them into a sequence of audio samples.

- The following are some of the ITU-T audio algorithms.
  - *G.711 Pulse Code Modulation (PCM)*
  - *G.722 Sub-band Adaptive Pulse Code Modulation* (SB-ADPCM).
  - *G.723.1*
  - *G.726 Adaptive Pulse Code Modulation (ADPCM)*
  - *G.722 Embedded Adaptive Pulse Code Modulation (EADPCM)*
  - *G.728 Low Delay Code Excited Linear Prediction* (LD-CELP)
  - *G.729*

Multi-frequency tones and CAS bits

- The User at the transmitter's side, detects and extracts dialed digits codes from multi-frequency tones, such as DTMF. It also extracts the ABCD CAS bits.
- At the receiver's side, it regenerates the multi-frequency tones from the received dialed digit codes and regenerates the ABCD CAS bits.

Facsimile

- Transmitter's side: It demodulate the facsimile signal and sends to the transmitting SSCS the demodulated image data and associated control signals.
- Receiver's side: It receives the image data and control signals from the receiving SSCS, it remodulates them into voiceband for transmission to the peer facsimile terminal.
- This demodulation/remodulation procedure provides a higher-fidelity transfer

The service specific convergence sublayer

- The SSCS provides a number of different services, such as audio service, circuit-mode service, frame-mode data service, dialed digits, CAS, alarms and state control information.
- An SSCS transmitter passes information from its *User* to CPS with no delay variation.

Type 1 and type 3 packets

- A transmitting SSCS passes data to CPS in CPS-packets (three-byte header and a payload which has a maximum length of 45 bytes).
- In AAL 2 SSCS for trunking, the CPS-packet payload is further structured to:
  - *type 1* packet (unprotected) or
  - *type 3* packet (fully protected).

  (Type 2 packets are to be defined.)

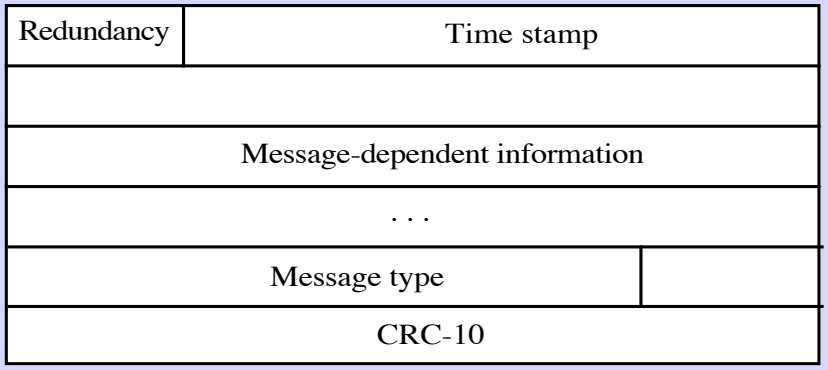
Type 1 packet

- In type 1 packets, the CPS-packet payload is simply made up of data without any additional information used for error detection, such as CRC or parity check.
- The maximum payload is now 45 bytes.

Type 3 packets

- The maximum payload: 43 bytes.
- Message type: a 6-bit field that contains a code to indicate the contents of the payload (dialed digits, ABCD bits, facsimile demodulation control data, alarms, and user state control operations).
- CRC: 10-bit field, $x^{10}+x^9+x^5+x^4+x+1$..

Common facilities for type 3 packets



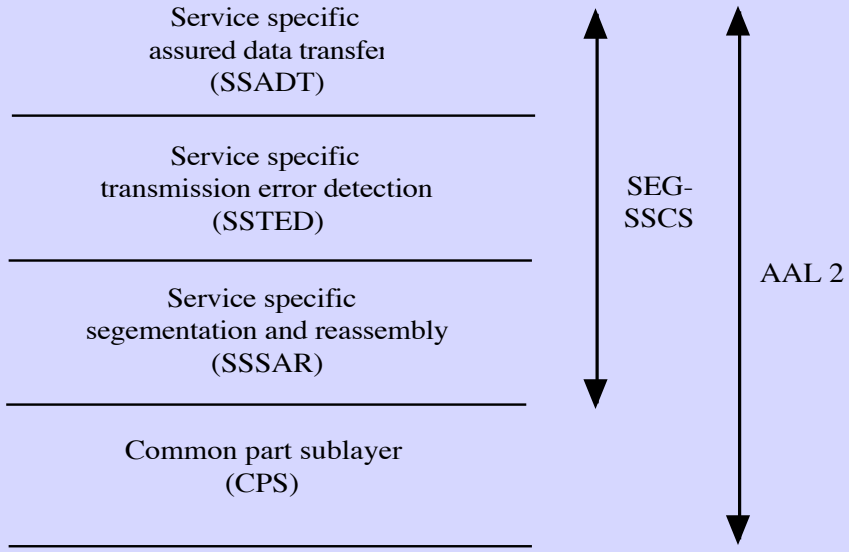
- This is a more detailed format of the type 3 packet used by the SSCS.
- It is triple redundant, i.e, it is transmitted three times with an interval that depends on the type of message

SSCS packets

- Different packets have been defined to transport the different types of data streams supported by the AAL 2 SSCS for trunking.
  - *Audio packet*
  - *Generic silence insertion description* (SID) *packet*
  - *Circuit-mode data at Nx64 Kbps packet*
  - *Dialed digits packet*
  - *CAS bits packet*

Segmentation and reassembly SSCS for AAL 2 (SEG-SSCS)

Using this service, it is possible to transport a packet with a size bigger than the maximum length of 45 bytes permitted in the payload of the CPS packet in an assured or non-assured manner



SEG-SSCS is sub-divided into the following sublayers:

- *Service Specific Segmentation and Reassembly (SSSAR),*
- *Service Specific Transmission Error Detection (SSTED) sublayer,*
- *Service Specific Assured Data Transfer (SSADT) sublayer.*

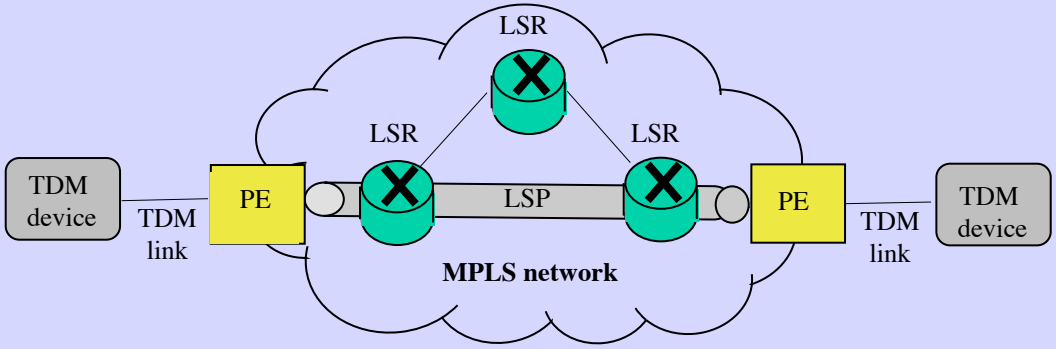
Voice over MPLS (VoMPLS

- MPLS provides quality of service on per connection basis as in ATM, and in view of this, it is a suitable technology for voice over packet.
- Implementation Agreements:
  - *TDM Transport over MPLS using AAL 1*
  - *I.366.2 Voice Trunking Format over MPLS*

TDM Transport over MPLS using AAL 1

- It defines a service which emulates a point-to-point TDM circuit, such as fractional DS1/E1 (nx64 Kbps), T1, E1, T3, and E3, over MPLS.
- It assumes that the TDM traffic to be carried over MPLS is already encapsulated in AAL 1 SAR-PDUs and it simply provides an efficient transport of the SAR-PDUs over an LSP

The TDM-MPLS reference architecture

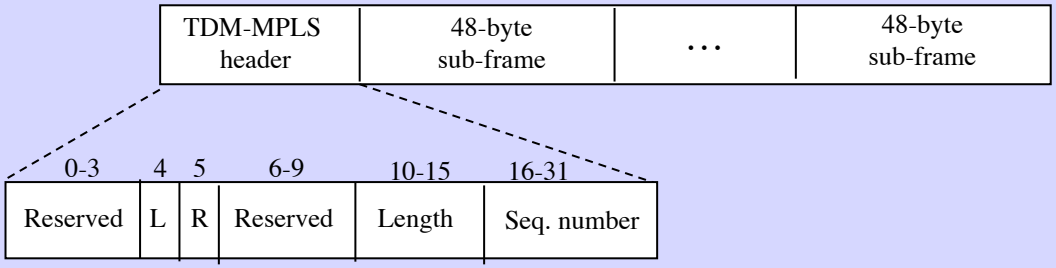


The *provider edge* (PE) provides a number of functions such as:

- Transport of fractional T1/E1, i.e. nx64 Kbps, or the entire signal of a T1, E1, T3, and E3, over an LSP.
- End-to-end preservation of the order of the TDM frames.
- Transparent transfer of CAS bits.
- A mechanism for the reconstruction of the TDM clocks.
- Transport of standard alarms between the two TDM devices.

TDM-MPLS frame

- The TDM traffic transmitted to a PE from the TDM device is first encapsulated using AAL 1.
- Many SAR-PDUs are then transported over MPLS to the destination PE in a TDM-MPLS frame

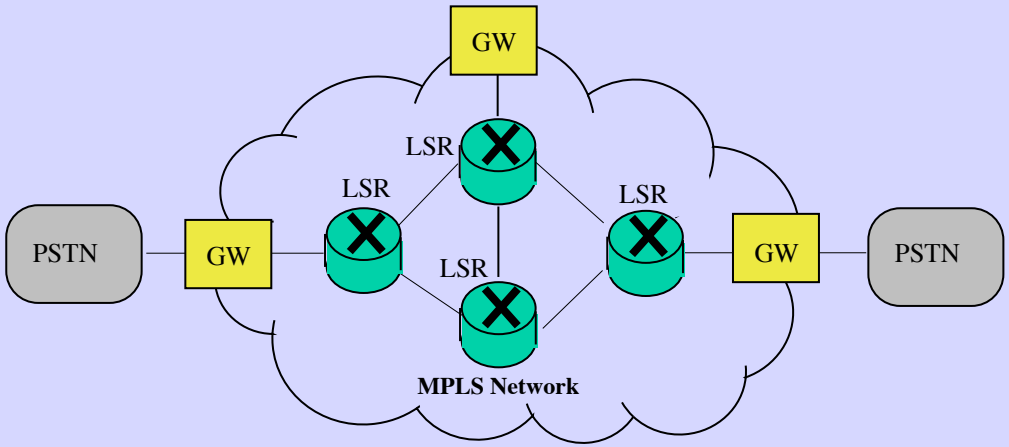


- *L bit*: Used to indicate physical layer loss of signal.
- *R bit*: Used to indicate that the source is not receiving packets at its TDM-MPLS receive port.
- *Length*: Used to indicate the length of the TDM-MPLS frame (header and payload) in case padding is employed
- *Sequence number:* Used to guarantee ordered frame delivery.

I.366.2 Voice Trunking Format over MPLS

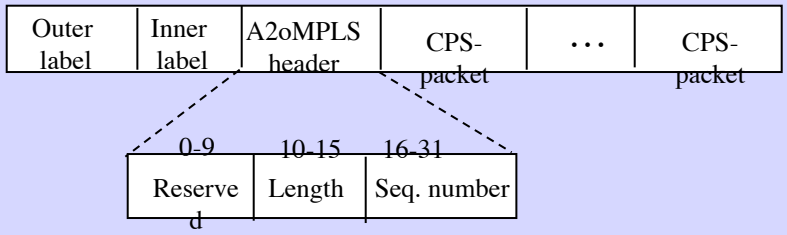
- It was defined to convey voice calls, voiceband data, such as facsimile and data transmitted over a modem, and fractional T1/E1 circuit-mode data.
- It assumes that the information to be carried over MPLS is the output of the AAL type 2 SSCS for trunking.
- In view of this it is also called *AAL2 over MPLS* (A2oMPLS)

The A2oMPLS reference architecture



The A2oMPLS frame structure

- In the AAL 2 CPS, the CPS-packets are packed into CPS-PDUs, and each CPS-PDU is carried in a separate ATM cell.
- In the A2oMPLS architecture, multiple CPS-packets can be placed onto the same frame, known as the A2oMPLS frame, and transported over an LSP.



- *Outer label:* Label used for MPLS routing.
- *Inner label:* An optional label used to increase the number of multiplexed voice calls onto the same LSP.
- *A2oMPLS header:* The header consists of a reserved field (bits 0 to 9), a *length* field (bits 10 to 15) and a *sequence number* field (bits 16-31) used to guarantee ordered frame delivery.
- *CPS-packets*: Same as in ATM AAL2