BASICS OF SWITCHING SYSTEM

i) Functions of Switching System

The switching office performs the following basic functions irrespective of the system whetherit is a manual or electromechanical or electronic switching system.

ii)**Identity.**The local switching center must react to a calling signal from callingsubscriber and must be able to receive information to identify the required destination terminalseize.

iii)**Addressing.**The switching system must be able to identify the called subscriberfrom the input information (train of pulses or multiple frequency depends on the dialing facility).The address may be in same local center or some other exchange. If the terminal or trunkgroup is busy, a suitable signal must be returned to the calling subscriber. If more than onefree circuit, particular one will be selected.

iv)**Finding and Pathsetup.**Once the calling subscriber destination is identified andthe called subscriber is available, an accept signal is passed to the switching system and callingsubscriber. Based on the availability, suitable path will be selected.

v)**Busy testing.**If number dialed by the calling subscriber is wrong or the calledsubscriber is busy (not attending the phone) or the terminal may be free (lifting the phone) butno response (not willing to talk or children handling), a switching system has to pass acorresponding voice message or busy tone after waiting for some time (status).

vi)**Supervision.**Once the path is setup between calling and called subscriber, it shouldbe supervised in order to detect answer and clear down conditions and recording billinginformation.

vii)**Clear down.**When the established call is completed, the path setup should bedisconnected. If the calling subscriber keeps the phone down first, the signal called clear forwardis passed to the switching system. If the called subscriber keeps the phone down first, a signalcalled clear backward signal is passed to the switching system. By clear signal, the switchingsystem must disconnect the path setup between calling and called subscriber.

viii) **Billing.**A switching system should have a mechanism to meter to count the numberof units made during the conversation. The cumulative number of units made for a particularduration by the calling subscriber is calculated. This information and if any should be sent tothe called subscriber.

Requirements of Switching System

All practical switching system should satisfy the following requirements for the economic useof the equipment's of the system and to provide efficient service to the subscribers. Depends onthe place (Rural or town, big town, city or big cities). The local exchange located, the serviceprovided to the subscriber may vary. Some important requirements are discussed briefly.

High availability. The telephone system must be very reliable. System reliability canbe expressed mathematically as the ratio of uptime to sum of the uptime and down time. The uptime is the total time that the system is operating satisfactorily and the down time is thetotal time that is not. In telephone switching networks, the availability or full accessibility ispossible if all of the lines are equally accessible to all incoming calls.The full accessibility is also defined as the capacity or number of outlets of a switch toaccess a given route. If each incoming trunk has access to a sufficient number of trunks oneach route to give the required grade of service is known as limited availability. The availabilityis defined as

$$A = \frac{\text{Uptime}}{\text{Uptime} + \text{down time}}$$

Also $$A = \frac{MTBF}{MTBF + MTTR}$$

where, MTBF − Mean time between failure

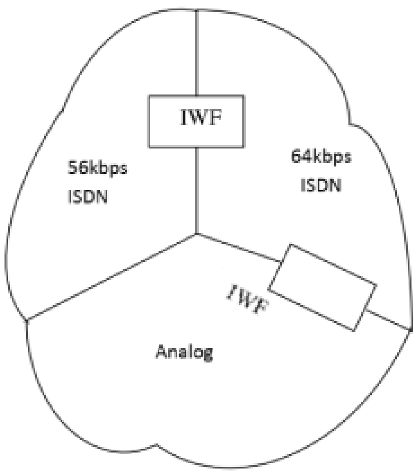
MTTR = Mean time to repair.

The unavailability of the system is given by

$$U = 1 - A < \frac{MTTR}{MTBF + MTTR}$$

INTERWORKING

ISDN interworking network based on three types of networks. i.e. the analog network, the 56kbps digital network for voice or data, and the ISDN with 64 kbps clear channel. Interoperability is achieved through interworking function (IWF) are different for different network and interface details.



IWF=Interworking Function

Fig.67: Coexistence of ISDN with other network

Separate reference points have been identified and their interface details have been defined to enable interworking with different networks.

Reference points	**Interworking With**
K	Analog or IDN (voice or data) when the IWF is implemented in ISDN
L	Analog or IDN when the IWF is implemented as part of existing network
M	Specialized service providers or value added networks(VANs)
N	Another ISDN
P	A specialized resource with ISDN

Interworking functions of ISDN with other networks having following objectives

1. Determine if the network resources of the nonISDN network are adequate to meet the ISDN service demand.
2. In case the network resources are inadequate, take suitable action to cancel the call, reroute the call or renegotiate with ISDN to be consistent with the available resources.
3. Map signaling messages two ISDN and nonISDN signaling systems.
4. Ensure service and connection compatibility, such as compatibility of call process tones, call failure indication and signal processing modules like echo canceller.
5. Provide transmission structure conversion, including modulation technique and frame structure conversion.
6. Provide error and flow control.
7. Collect data for billing and charging.
8. Coordinate operation and maintenance procedures to be able to isolate faults.
9. Enable interworking of the numbering schemes.

The above interworking functions point to need the parameter exchange among the existing nonISDN terminals, ISDN terminals, and the strategically located IWF modules.

Numbering Interworking

The international PSTN numbering scheme used a 12 digit numbers with a two part format of country code and national significant number. Other ISDN also uses 15 digit numbers.ISO defined its own numbering scheme in the context of OSI reference model. However two basic methods for interworking between the ISDN numbering plan and other have been proposed:

- ➔ Two-stage or port method
- ➔ Single stage or integrated numbering method.

In the port method the numbering plan of the network to which the called party is attached is treated as calling party's network and a two stage dialing is used.

The main disadvantages of two stage dialing are:

- ➔ The user dials two sets of numbers.
- ➔ Dialing is dependent on the relative numbering plans of the called and calling terminal.
- ➔ A delimiter or pause is necessary between two stages e.g to obtain the dial tone of second network.
- ➔ The dialing becomes even more cumbersome if more than two networks are evolved in establishing a connection.

An important advantage of two stage scheme that is simple to implement and is universally acceptable.

In a single stage or integrated numbering scheme, a prefix is used to identify a particular network and thereafter the numbering scheme of the destination network is dialed.

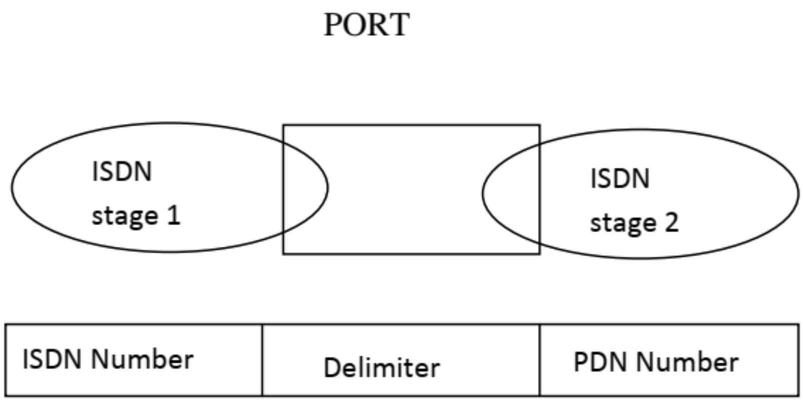


Fig.68 (a): Two stage dialing [1]

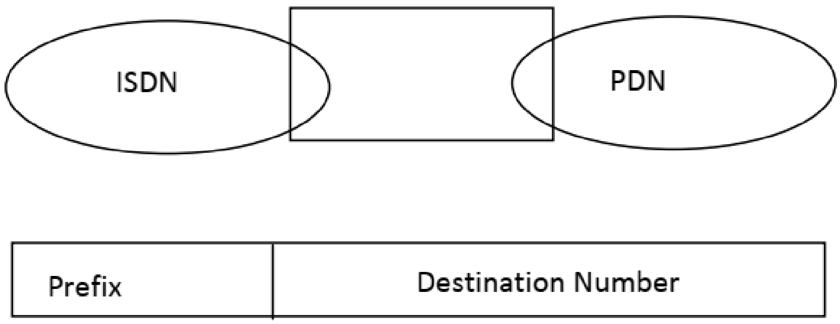


Fig.68 (b): Single stage dialing [1]

BROADBAND ISDN (B-ISDN)

B-ISDN provides the needs (High speed and large data handling) of the next generation technology. B-ISDN is a digital service with speed above 1.544 Mbps. The original ISDN is called narrow band ISDN (N-ISDN). B-ISDN uses fiber at all levels of telecommunications. BISDN provides two types of services. They are interactive (conversational or messaging or retrieval). The interactive service is bidirectional. The distributive services are unidirectional (with user control or without user control). Several forms of B-ISDN exist. Some are listed below:

Frame relay service:-Frame relay is considered to be a B-ISDN service. Frame relay is a packet switching protocol service offered by telephone corporations to replace the X-25 protocol. It is a WAN network.

Switched Multimegabit Digital Service (SMDS):- SMDS is a digital service that provides a high speed digital path. The transport speed of SMDS is usually 155 Mbps.

ATM:-The transport speed of most ATM applications is 155 Mbps. B-ISDN uses the same functional groupings of N-ISDN. But, named as B-NT1, B-NT2, B-TE1, B-TE2 and B-TA. B-ISDN uses the same R, S, T and U reference points. However BISDN uses three different access methods to satisfy the user needs. They are symmetrical 155.52 Mbps, asymmetrical 155.520 Mbps and symmetrical 622.080 Mbps.

VOICE DATA INTEGRATION

Generally in segregated infrastructure, we have collection of different types of networks; like packet switched circuit switched, no switched and so on. Depending on the nature of the traffic of the service, the service is carried on. For example voice traffic is carried on circuit switched network, whereas data traffic is packet switched network. In this topic we have considered about the traffic i.e. of two types: digitized voice and data. In this section the part is discussed about voice data integration on a single channel.

Table: Parameters relating to Digitized voice and data traffic

Digitized Voice	Data
Periodic busty in nature	A periodic busty in nature
Fixed length bursts	Variable length bursts
Small packet size	Large packet switch
Packetisation time critical	Packetisation time is not critical
Hard bound on delay	Soft bound delay
Hard bound on variance of delay	soft bound on the variance of delay
Loss of parts of speech acceptable	Loss of part of data unacceptable
Low overhead as there is no error recovery	High overhead due to error detection and recovery

Data traffic is aperiodic whereas voice traffic is periodic. The traffic for digitized voice may be considered to be busty, through periodic as the transmission may be organized on sample by sample basis i.e. one byte in every 125 µs.If packetized voice transmission is used, and then the packet size is to be small to maintain the real time characteristics of the service. Large packet sizes would lead to be broken speech transmission. For similar reason the packetisation time is critical for voice packets. Voice packets should be reached the destination within the specified time or else speech would discontinuous. A basic model for integrating voice and data is to considered a TDM frame that is capable of carrying both circuit and packet switched data. A portion of frame Fc bits is occupied by the circuit switched data and the remaining portion Fp bits by packet switched data as shown in figure below.

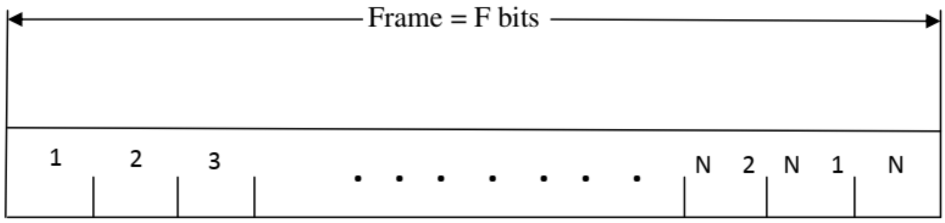


Fig.69: An integrated TDM network [1]

Circuit switched traffic= Fc bits

Packet switched traffic= Fp bits

Fc+ Fp=F

A number of circuit switched sources may constitute the Fc bits of the frame. Each source may require different bandwidth and have different arrival time and holding time characteristics. A TDM frame may considered containing N slots of b bits each, as a group of b bits likely to be integral in any traffic.e.g. Voice samples of 8bits or a byte in data.

F=Nb slot ,One slot may be considered as a basic bandwidth unit(BBU) .A source is assigned one or more BBU's depending upon the bandwidth requirements. A circuit switched source is blocked if adequate no. of BBUs are not available for the assignment and a packet switched source is queued awaiting free BBUs. For voice data integration we restrict the circuit switched traffic to voice. i.e. all sources have equal bandwidth requirement of one BBU and the same arrival time and the holding time characteristics. Similarly all packet switched sources may be considered to have the same arrival and service time characteristics for simplicity.

Voice data Integration schemes followed by may be placed under two characteristics:

- ➔ Random allocation of BBUs
- ➔ Contiguous allocation of BBUs

In random allocation, any available BBU is allocated to either voice or data using some criterion. The usual criteria include First-come-first-serve basis (FCFS) and pre-emptive scheduling. In contiguous allocation schemes attempts are made to gather together all BBUs allotted to voice and so also BBUs allotted to data traffic. The contiguous allocation schemes can be classified as:

- ➔ Fixed boundary scheme
- ➔ Movable boundary scheme

In Fixed boundary scheme, the N slots in the frame are divided into two parts containing N1 & N2 =N-N1, slots respectively as shown figure 65(a).

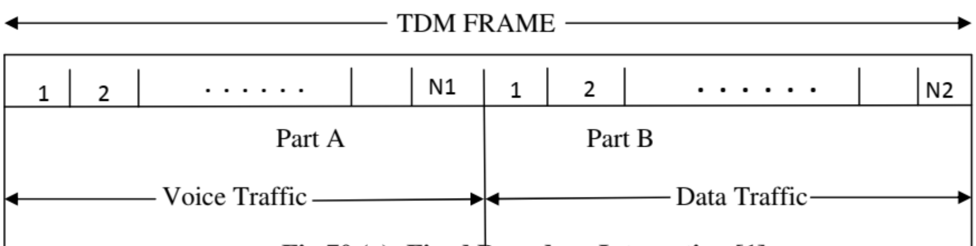
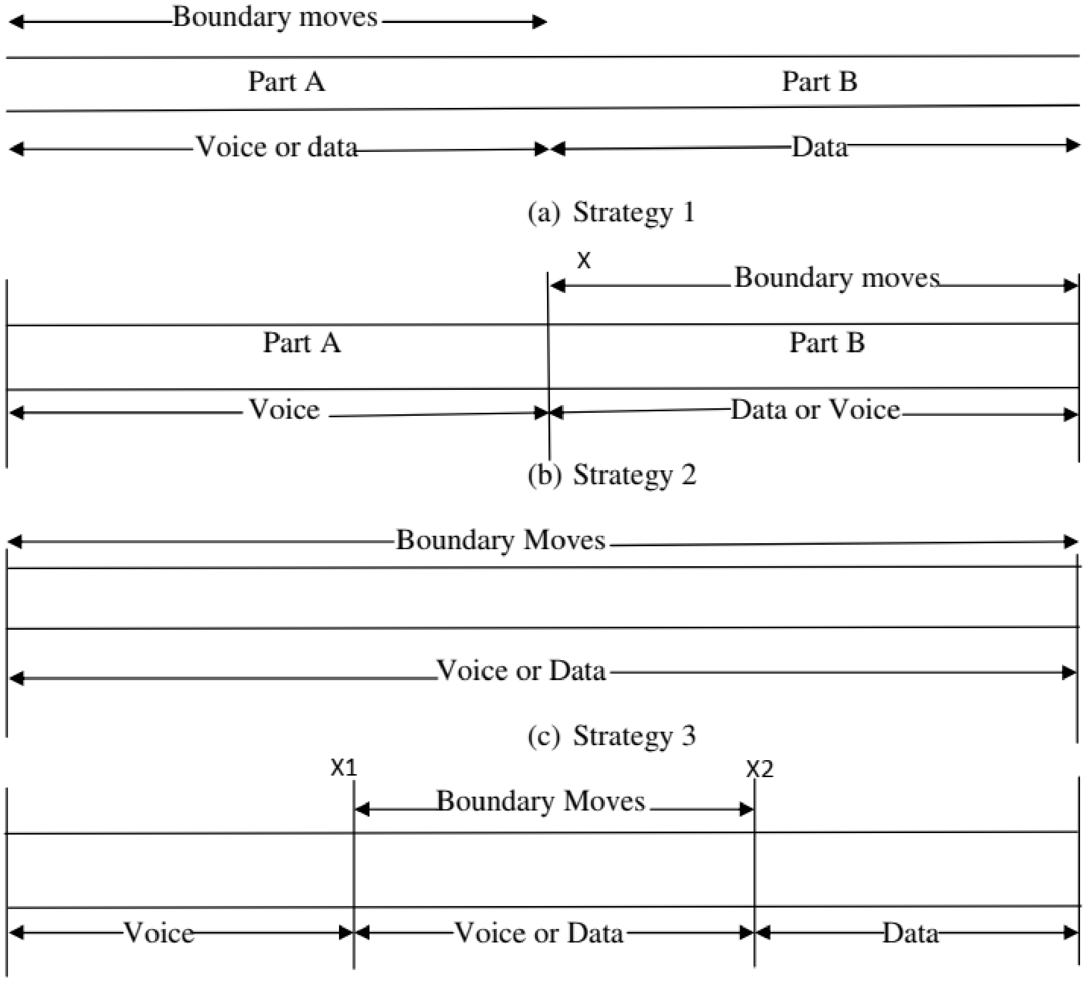


Fig.70 (a): Fixed Boundary Integration [1]

It has two parts part A for **voice traffic** and part B for **data traffic**. Voice traffic is always scheduled in part A, and if there are no free BBUs in that part, the call is blocked, even though free BBUs may be available in part B. Similarly data traffic will be queued if there are no free BBUs in part B,even though free BBUs available in part A. The performance is measured in terms of blocking probability for voice sources and mean delay for data sources.In movable boundary scheme, four strategies are possible as shown in figures below.



(d) Strategy 4**Fig.70 (b): Movable Boundary Schemes**

In strategy 1 and 2, there is a national boundary shown at X. In the former data traffic is allowed to use free BBUs. If a voice source arrives, then a data source occupying the voice area is pre-empted and the BBUs are allotted to the voice source .In this sense the boundary moves back and forth to the left of X. If all the BBUs in part A occupied by voice sources ,further voice arrivals are blocked even if free BBUs are available in part B. This strategy gives the same blocking probability performance as the fixed boundary scheme for voice sources;but it provides better delay performance for data sources.

In strategy 2, the boundary is allowed to move according to figure shown above, thereby providing additional bandwidth to voice sources when available. Consequently if the voice sources hold up BBUs in the data traffic for a long time, the delay performance for the data traffic may be affected adversely.

In strategy 3, the boundary shifts dynamically throughout the frame, depending on the traffic load. Pre-empting may be permitted in this strategy for voice traffic. In strategy voice may be completed domination the channel, hence forth some researchers referred to **strategy 4** as shown in figure, where a minimum bandwidth allocation is always assured for both voice or data traffic. Such traffic is also known as **min-max scheduling** as there is a minimum and maximum bandwidth bound for either of traffic.

REFERENCES

1. Duhnkrack, George (April 1960). The Electronic Switching System. Bell Telephone Laboratories, Incorporated.
2. Allstot, David J. (2016). "Switched Capacitor Filters". In Maloberti, Franco; Davies, Anthony C. (eds.). A Short History of Circuits and Systems: From Green, Mobile, Pervasive Networking to Big Data Computing (PDF). IEEE Circuits and Systems Society.
3. Floyd, Michael D.; Hillman, Garth D. (8 October 2018) [1st pub. 2000]. "Pulse-Code Modulation Codec-Filters". The Communications Handbook (2nd ed.). CRC Press.
4. High-Performance Communications Networks", Second Edition, Jean Walrand and Pravin Varaiya, Morgan Kaufman Publishers
5. Softswitch : Architecture for VoIP (Professional Telecom)Dec 10, 2002 by Frank Ohrtman