Chapter VIII

FUNDAMENTAL TECHNICAL PLANS

Contents

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Purpose of this chapter

As their name implies, fundamental technical plans are used as references which define the technical standards which are to be achieved, and maintained, by the operator over the long-term planning period. Most of the quantitative standards to be achieved are defined in CCITT / ITU-T and CCIR / ITU-R Recommendations, these being mentioned in the guidelines when appropriate. The guidelines in this chapter are intended to assist planning personnel in the preparation of the fundamental plans required as part of the strategic planning process, in particular to meet the requirements for totally digital networks.

Outputs expected

- Fundamental Technical Plans, updated according to the latest ITU Recommendations and prepared according to these guidelines;
- Technical documentation for future use by the responsible planning section in the further up-dating of these plans.

Inputs required

- All the relevant ITU-T and ITU-R recommendations together with the manuals published by the ITU;
- The outputs listed for Chapter II (background information on existing network);
- The guidelines contained in this chapter.

Chapter VIII

FUNDAMENTAL TECHNICAL PLANS

8.1 Numbering systems

8.1.1 General aspects

Numbering systems for telephone networks can be characterised by two basic features:

- open or closed numbering systems;
- variable or uniform (fixed) numbering length.

A network with a closed numbering system consists of one single numbering area; one dialling procedure applies to calls within the national network, i.e. by dialling the subscriber number.

A network with an open numbering system consists of a certain quantity of numbering areas, each area having been allocated a trunk code and, within the areas, a closed numbering system is applied with one dialling procedure; for calls to other numbering areas, a different dialling procedure is applied by dialling a trunk code followed by the subscriber number.

The trunk code consists of a trunk prefix followed by a number of digits characterising the numbering areas. This number of digits depends on the size of the numbering areas.

In some networks, the trunk code can be one or two digits for numbering areas covering multi-exchange metropolitan networks or large regional networks, while for other networks it can be three or even four digits, for numbering areas covering small groups of exchanges or even single exchange local networks.

A uniform or fixed numbering length is generally applied within a closed numbering system (either a closed numbering system on national level or on the level of numbering areas in an open numbering system).

If a differentiation is applied, then it is only on the difference in numbering length of one digit, and to be identified by analysis of the first or first two digits only, for the sake of convenience of both subscribers and equipment. In general, such a differentiation of numbering length is only applied during a transitional stage, when the numbering capacity in a network or in a numbering area will be increased to meet the demand in such a network.

In a system with variable numbering length, the differentiation of number length is accepted as a principle; however, this applies only to open numbering systems where the national significant number (= trunk code without prefix + subscriber number) may vary from for example 7 to 9 digits (2+5, 2+6 or 2+7). However it is recommended that a fixed number length be adopted whenever possible.

Regarding standardisation, CCITT recommends "0" as trunk prefix and "00" as international prefix. As far as the national significant number is concerned, it is recommended to limit and preferably to have less than 12-n digits in the national numbering system (n = number of digits of the country code, generally 2 or 3 digits).

These requirements are included in CCITT Recommendation E.163 and they were applicable to the Public Switched Telephone Network (PSTN). A new recommendation E.164 will apply for the ISDN era and numbering systems have to be based on this recommendation.

Generally, a range of short codes is reserved within a national numbering plan for special services. There are no specific CCITT recommendations about these codes, but agreements may exist in different world regions about the application of certain numbering arrangements for special services. In view of the large number of special services now available it is recommended that a 3-digit code be adopted (1xx).

Finally, certain special access codes may be required for special networks such as mobile telephone or free-phone or special charged services. Such access may be realised by either the use of a free level in the (local) area numbering, or the use of a free trunk code, or by taking a spare code of the special service code numbering range.

Examples of international free-phone service codes are given in CCITT Recommendation E.152.

8.1.2 Review of the existing numbering system and formulation of future requirements

The review of the actual numbering system should include an analysis of the availability and the current requirements for special service codes and access codes to particular networks and services.

Then, for each numbering area, an assessment has to be made of its actual available spare numbering capacity and numbering saturation problems, if applicable.

The future requirements have to include the following aspects:

- the increased requirement for numbering capacity, in view of the increased subscriber demand, including requirements for direct-dialling-in to PABX-subscribers and ISDN, preferably based on long-term forecast for subscriber demand development over periods in the order of 30 to 50 years;

- the expected long-term requirements for access codes to specialised networks, special services and free-phone or other special charged services;

- The ISDN-numbering plan, included in CCITT Recommendation E.164, which provides for international ISDN numbers up to 15 digits. It is expected that the ISDN and the PSTN will be mixed and the new E.164 numbering plan will also apply to the former PSTN.

  For addressing terminals in an ISDN-bus (S-bus) two different methods can be envisaged:

  1. With address information included in the Multiple Subscriber Number (MSN), which is part of the normal ISDN-number.

  2. By means of sub-addressing, which consists of an additional field of maximum 20 octets (40 digits), and this is separate from and in addition to the ISDN number.

For calls from PSTN subscribers (analogue telephones) to ISDN telephones, it is necessary that the address information is included in the ISDN number (MSN). If the MSN method is used, then additional numbering capacity has to be reserved for ISDN terminations.

The timetable for coordinated implementation of the full capability of the numbering plan for the ISDN era is indicated in CCITT Recommendation E.165, the time having been set for 31 December 1996.

The most difficult issue is certainly the long-term forecast of the subscriber demand including the requirements for direct-dialling-in to PABXs and ISDN connections, and it is clear that only a global estimate of these requirements can be made.

Normally, these long-term demand estimates are included in the development plan in the section dealing with demand forecasting, but if a fundamental numbering plan revision is required independently from the preparation of a master plan, then it will be necessary to develop such a forecast, which can be based on expected development of the telephone or terminal density, household penetration, etc. (see also § 8.2.4 for an empirical method).

The allowance to be provided for additional numbering capacity for direct in dialling to PABXs and ISDN multiple subscriber numbers can be considerable for the purpose of estimation, sometimes in the order of 50% of the "normal" requirements for numbering capacity.

8.1.3 Numbering plan modifications

Modifications in numbering plans can basically be classified as follows:

Under the first category can be mentioned:

- modification of prefixes;
- revision of trunk codes;
- provision of special codes and/or numbering series for free-phone and other special charged services;
- overall changes of the numbering length of the national number in order to harmonise the national numbers and to increase the overall numbering capacity of the system;
- changes to accommodate other networks and ISDN - type services.

The second category contains the individual modifications of particular numbering areas, within the limits of the national numbering system (with variable numbering length), with the aim of increasing the capacity of these numbering areas.

As far as the national numbering plan structure is concerned, the modification of prefixes is desirable in networks where the recommended prefixes "0" and/or "00" are not yet being used for trunk traffic and international traffic respectively. This may require a rearrangement of other codes already starting with "0" or "00".

A revision of trunk codes may be required for various reasons, one of them being the need to provide more numbering areas in expanding networks and/or to reallocate trunk codes in such a way that short trunk codes (one digit) are allocated to one or two big numbering areas (including metropolitan networks), two-digit trunk codes to medium-sized numbering areas and three-digit trunk codes to small numbering areas, with the aim of maintaining a uniform numbering system for the whole network.

Another reason could be the need to provide special trunk codes for special networks (mobile networks for example) or for free-phone and other special charged services.

The local numbering within numbering areas may have to be changed if special service codes, which up to now were included in numbering series outside the local numbering scheme, will have to be allocated with short codes starting with "1", due to regional agreements on this (and other) subjects.

The same applies if free-phone and other special charged services will have codes allocated from the local numbering range.

In both cases this may entail rearrangement of local subscriber numbers in the different numbering areas.

Finally, harmonisation of the numbering length for the entire network may also entail adjustment of subscriber numbers by adding one or a few digits to existing local subscriber numbers.

As far as the second category is concerned, the increase of the numbering capacity within a certain numbering area will be more convenient for both equipment and subscribers, if such an increase can be achieved by adding a free level (value) of the first digit to the existing numbering. In this case it will be easy for the equipment, by the analysis of the first digit, to determine if the selected number is a new (correct) or an old (faulty) number, and give an announcement to the calling subscriber in case of selection of an old number.

Last but not least, for any proposed changes in the numbering, it is of vital importance that the subscribers are adequately informed about these changes, for important changes by circulars, by publications and announcements in the different types of media and, if possible by information in the telephone directories.

Announcement machines should be used initially as much as possible to inform subscribers dialling old (wrong) numbers, which eventually can be changed to special information tone.

In general, it is essential that spare numbers be reserved to accommodate future changes.

8.1.4 Equipment requirements and implications

Numbering capacity has to be distributed in a systematic way in order to allow for the analysis of the first (few) digits of a number only, for the purpose of routing and charging.

As far as digital exchanges are concerned, one or more blocks of 10000 numbers are allocated to such exchanges, with the possibility to distribute blocks of 1000 numbers to remote switching units (RSUs) connected to these exchanges and to small stand-alone units.

The digital exchanges are very flexible regarding numbering and do not normally create any problem concerning number analysis and numbering length as required in Recommendation E.164 (international number maximum 15 digits). However, difficulties have been experienced due to the fact that the actual programming of supplied exchanges limits this flexibility. It is therefore recommended that the wording of the switching procurement specification should ensure that future numbering changes shall be possible without any changes whatsoever to the hardware or software of the exchange.

However, problems may occur with existing electromechanical exchanges which may have digit storage capacity tailored to the existing numbering plan, and where it may be difficult or even impossible to accept one or more additional digits due to the expansion and/or modification of the existing national numbering plan.

The same applies to electromechanical step-by-step exchanges, which are inflexible regarding local numbering, but flexible in national numbering (open numbering system) and also in international numbering.

In such cases, it may be necessary to adopt interim numbering arrangements in numbering areas with the existing numbering length for the old electromechanical-exchanges and with the new numbering length for the new (digital) exchanges. An example is given in the next paragraph.

Another problem related to exchanges with limited digit storage capacity, although they maybe able to meet national and even current international numbering requirements (maximum 12 digits), is the new requirement for E.164 numbering systems, and some interim arrangement (two-stage selection) may be envisaged for subscribers connected to those exchanges, if such a situation might occur.

8.1.5 Example of numbering change

The following is a typical example of the numbering changes which have to be introduced when analogue exchange units having limited number capacities have to be replaced by digital units due to expansion.

a) Existing numbering scheme:

5-digit numbering type ABCDE, with the following allocation of first-digit (A)-levels:

A = 0	prefixes
= 1	special service codes
= 2 - 7	six existing 5-digit analogue (EM) exchange units
= 8,9	spare

b) Installation programme:

Phase 1 : replacement of 3 analogue exchange units (A = 3,4,6) by digital exchange units, initial numbering capacity 10000 lines each, installation of additional digital switching units : 40000 lines;

Phase 2 : replacement of 3 remaining analogue exchange units (A = 2,5,7) by digital exchange units (10000 lines each), extension of existing digital exchanges and installation new digital exchange : 80000 lines.

c) Change to be introduced:

existing (5 digits)	phase 1 (5/6 digits)			phase 2 (6 digits)		
2xxxx (EM)	2xxxx	(EM)		82xxxx	(D)	new
3xxxx (EM)	83xxxx	(D)	new	83xxxx	(D)	
4xxxx (EM)	84xxxx	(D)	new	84xxxx	(D)	
5xxxx (EM)	5xxxx	(EM)		85xxxx	(D)	new
6xxxx (EM)	86xxxx	(D)	new	86xxxx	(D)	
7xxxx (EM)	7xxxx	(EM)		87xxxx	(D)	new
A = 8,9 : spare	80xxxx	(D)	new	80xxxx	(D)	
	81xxxx	(D)	new	81xxxx	(D)	
	88xxxx	(D)	new	88xxxx	(D)	
	89xxxx	(D)	new	89xxxx	(D)	
	A = 9		spare	90xxxx	(D)	new
				91xxxx	(D)	new
				.....		
				97xxxx	(D)	new
	AB = 82,85,87 reserved for replacement analogue exchanges			spare : AB = 98, 99 A = 3,4,6		
	wrong number announcement on A = 3,4,6			wrong number announcement on A = 2,5,7		

NB. Wrong number levels should remain blocked for a few years.

8.1.6 Example of a closed numbering scheme

The following is a résumé of the numbering scheme being adopted by the Norwegian telecommunication authority. It consists of uniform 8-digit subscriber numbers.

a) The 1 - series is reserved for special numbers allocated to public and emergency services etc. (100, 101, 102,....)

b) The 2, 3, 5, 6 and 7 series of numbers are allocated to different geographical areas.

c) The 4 - series is retained as a spare series for the future.

d) The 8 - series is allocated to free-phone services etc.

e) The 9 - series is reserved for the cellular radio network.

In the Oslo region for example the series 21, 22 and 23 are reserved, all numbers in the 22 range being allocated to TELENOR, the principal operator, giving a capacity of 1 million numbers. As this series is close to its maximum utilisation, the new series 230 and 231 will probably be allocated to increase the capacity to 1.2 million. Any new operator in this region would be allocated the series 232 to 234. Thus there is sufficient capacity to meet future requirements by the allocation of the series 235 and beyond.

In the provinces, Vestfold for example has been allocated the 33 - series of numbers, TELENOR at present using the 330, 331, 333, 334 and 337 series. The remaining allocations 332....339 are not yet used. Any new operator in this region will be allocated numbers in the remaining free series.

Mobile operators, as mentioned above, are allocated 9 - series numbers (9xxx xxxx). There are two mobile operators in Norway; TELENOR and NETCOM. The series 900 to 907 have been allocated to TELENOR, providing 800,000 numbers, while NETCOM has been allocated the series 920 to 927, also providing 800,000 numbers. If more numbers are required the series 928 and 929 will be used.

The cellular network operators are allowed to implement their own numbering schemes within the allotted series - as will other operators after 1998 - but are required to report on how their numbering plan will be implemented to the regulator, the Telecommunication Authority.

8.1.7 References

1. CCITT Blue Book Volume II, Fascicle II. 2, Geneva 1989:

   - Recommendation E.160 - Definitions relating to national and international numbering plans;
   - Recommendation E.163 (Q.11) - Numbering plan for the international telephone service;
   - Recommendation E.164 (I.331, Q.11 is) - Numbering plan for the ISDN era;
   - Recommendation E.165 (Q.11 tour) - Timetable for coordinated implementation of the full capacity of the numbering plan for the ISDN era (Recommendation E.164);
   - Recommendation E.166 - Numbering plan interworking in the ISDN era.

2. CCITT Recommendation Q.81.2 & 8 - Multiple Subscriber Number (MSN) - Sub-addressing (SUB).

3. CCITT Manual: Economic and Technical Aspects of the Choice of Telephone Switching Systems, ITU, Geneva, 1981 (GAS 6 Handbook).

8.2 Routing and switching plan

Routing and **Switching** are two important subjects in the activity or process called **Network Dimensioning and Configuration**. These two subjects appear both as an input to as well as an output from the process.

8.2.1 Routing

Routing as **input** describes the prescribed routing rules on the different levels of the network and in the different particular service areas. One extreme would of course be to give no rule at all, which would mean that no restrictions were put on the planners, i.e. the actual routing would be obtained in the network dimensioning process according to other criteria, for example. as a result of minimising the total network cost. This latter case is however very unusual. More common is that routing rules are given, but they should always leave as much freedom as possible to the network planner, since all forms of prescribed rules really are constraints for the planning and can only result in a more expensive network solution.

The routing rules determine how the traffic is routed from one subscriber to any other. A large number of traffic cases may occur, concerning for example:

1 Geographical level
Rural
Metropolitan
National

2 Network - Teleservice centred view
PSTN - Public Switched Telephone Network
PSPDN - Packet Switched Public Data Network
CSPDN - Circuit Switched Public Data Network
PLMN - Public Land Mobile Network

3 Network - Bearer service and technological centred view
Analogue network
Digital overlay network
N-ISDN - Narrowband ISDN
B-ISDN - Broadband ISDN

4 Switching technology
Analogue switching
Digital switching, 64 kb/s
ATM - Asynchronous Transfer Mode

5 Traffic type
Local traffic
Long-Distance traffic
International traffic

6 Traffic direction
Outgoing traffic
Incoming traffic
Internal traffic

7 End user
Single main line
2B+D ISDN line
30B+D ISDN line
PABX
LAN

Many different routing cases may be defined from the above list by combination of alternatives under each item 1-7, and the list relevant for a particular country may be more complex, longer, and in any case different from the one shown above. Therefore, the principle of **Simplicity** in network design is also of value concerning routing rules.

Telecommunication networks tend to be more and more complex, Figure 8.1 showing an example of the possible future situation in a typical telecommunication area.

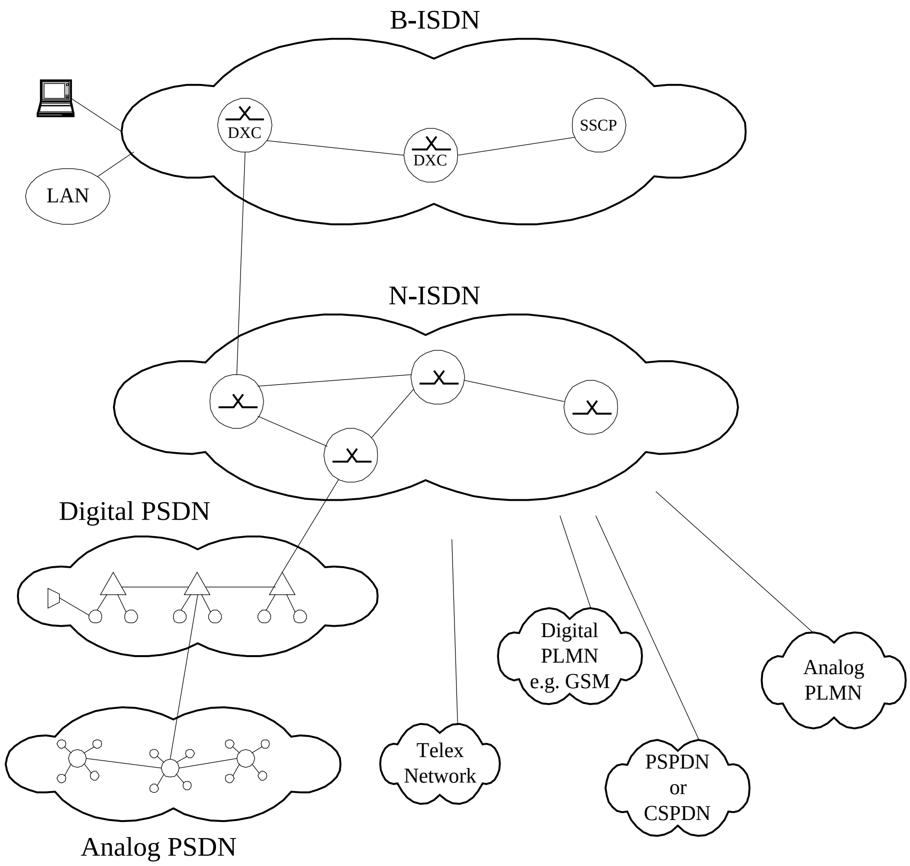


Figure 8.1: Future integration: cooperating networks

Routing as an **output** from the network dimensioning and configuration process shows how the routing rules given as input have been implemented in each individual case, utilising the possibilities of the rules and obeying their restrictions.

The planning group should therefore define the routing plan in two different ways:

1. The **Routing Rules** or principles which are prescribed for the various levels and cases, and which are the input to network dimensioning;

2. The **Detailed Routing** for each individual area and case that is the result of network dimensioning.

This means that as the network develops, the **Routing Rules** should always be obeyed, but the **Detailed Routing** can, and should, change over time. For an ordinary PSTN, the following list provides an example.

Routing rules should include:

- The stated planning in terms of overall policies towards the maximum number of links in a connection, direct routes, alternative routing, transit routing, hierarchical routing and the implications of any switching, signalling, numbering and transmission conversion plans;

- The maximum number of transit links allowable with particular combinations of switching and transmission;

- The dimensioning principles for all route classes:
  a) subscriber lines including remote subscriber units;
  b) routes between local and primary exchanges;
  c) direct routes;
  d) high usage routes;
  e) alternative routes;
  f) final routes.

Detailed Routing for each local area should cover:

a) subscriber lines;
b) remote subscriber units;
c) direct routes between local exchanges within the area;
d) direct routes to exchanges outside the area;
e) routes between local exchanges and the primary exchange;
f) special routes such as operator services, emergency access;
g) transit routing;
h) alternative routing and the security implications;
i) high usage routes;
j) overflow routes.

Detailed Routing for the national network:

a) routes between primary exchanges;
b) routes to secondary exchanges and between them;
c) all other routes in the backbone structure;
d) high usage routes;
e) overflow routes;
f) alternative routes and the security implications;
g) direct routes across international boundaries;
h) access to the international exchange.

Finally, the routing plan should be summarised by using a hierarchical diagram as shown in Figure 8.2, and by describing the paths taken by typical calls in various traffic conditions.

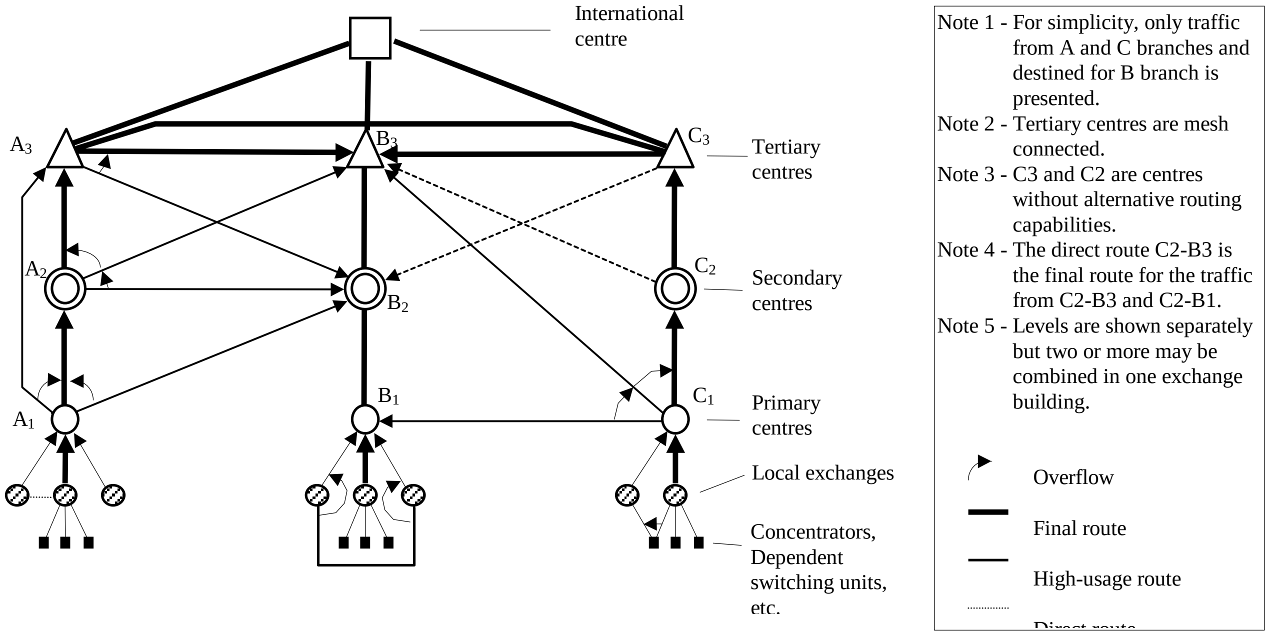


Figure 8.2 : Example of a hierarchical routing plan

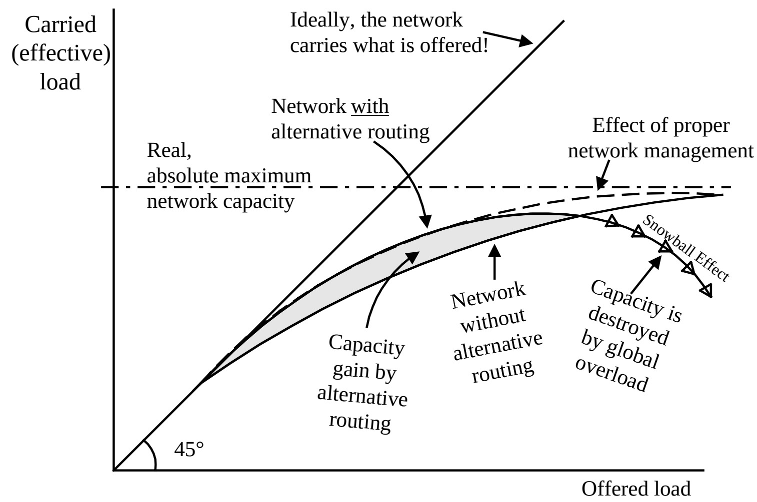
The dominating type of network is still the **Hierarchical Network**, where several levels of switches are defined. When **Automatic Alternative Routing** is correctly applied in a hierarchical network, the network efficiency is improved for two reasons:

- Direct routes are limited in size, so that the traffic utilisation increases due to a higher degree of competition among the calls. On the other hand, the direct routes can then not handle all the traffic offered to them, so a part of the traffic must be directed through other, alternative routes in the network;

- Overflow traffic from many such traffic cases may share the same alternative route. The total traffic on an alternative route may therefore be quite large, which circumstance increases the traffic utilisation also on that route.

The total utilisation in the network may therefore be increased by automatic alternative routing. This increased utilisation can be used in two ways: either by saving channels and thus decreasing the network cost, or by improving the grade of service. Another quality is also obtained in that the alternative routes are common reserves to several traffic cases. Furthermore, the reliability is increased since the traffic always has at least two possible paths.

There are also drawbacks: one is that an average call now uses more than one link; another is that the network becomes more sensitive to global overload compared to simple, non-alternative routing. These two properties may create a snowball effect in situations with serious overload: the high load causes more calls than before to use more links, which means that the network occupancy increases, which in turn causes even more calls to use even more links etc., so the network occupancy may rapidly increase, with the result that a considerable part of the network's nominal capacity may be absorbed and the useful capacity may be severely decreased. (See Figure 8.3).



If the network is overloaded, it will take some time to come back to normal load again, after the offered load has decreased :

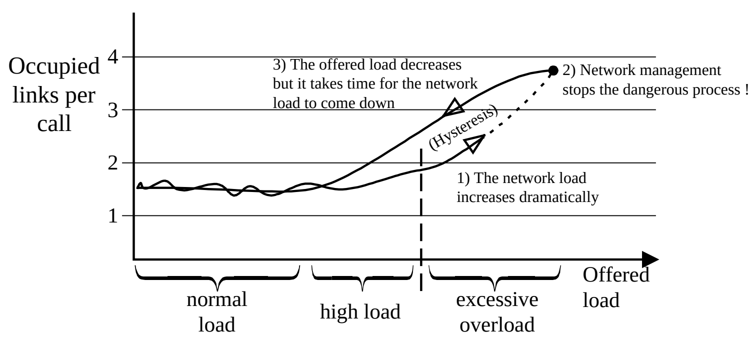


Figure 8.3 : Traffic properties of alternative routing networks

This situation can only be relieved if network management actions are taken, for example by cancelling the overflow possibilities for new calls, until the network occupancy has returned to more normal levels.

It is possible to employ other routing strategies besides hierarchical alternative routing. A large class of such strategies is used in **Non-Hierarchical Networks**, where the sequence of first and alternative choices is not only unique for each pair of originating and terminating switches in the network, but may also change from time to time. This means that each exchange must have a whole set of routing tables, and which table to use when a call arriving at a particular switch is to be transited further depends on the identity of the specific origin/destination pair. Each exchange may be ordered to use a completely new set of routing tables from time to time. If this process is clock controlled, the routing strategy is **Dynamic**; if the process is controlled by the actual network load at various points, the strategy is **Adaptive**.

Since adaptive routing to a certain extent adapts to the actual network load, is to be expected that less extra network management will then be needed. However, the precise way of adaptation is dependent on calculating algorithms which are constructed by traffic engineers, so that no adaptive system can be absolutely perfect. It is always possible that new and unforeseen combinations of circumstances will occur.

Another factor is that adaptive routing requires a somewhat sophisticated arrangement with data links from each switch to a centrally placed main frame computer. When used, non-hierarchical routing is usually first implemented on the highest levels in the national network. Instead of having for instance two switches on the highest level and five switches on the next level hierarchically interconnected, all seven switches may be fully interconnected and work non-hierarchically. A connection between any pair of switches may then have a great number of possible paths via other switches.

A routing arrangement which gives a high degree of reliability and is especially suitable for large metropolitan areas is to have full interconnection between all tandem exchanges, letting each local exchange have two entrances to the tandem level, as shown in Figure 8.4.

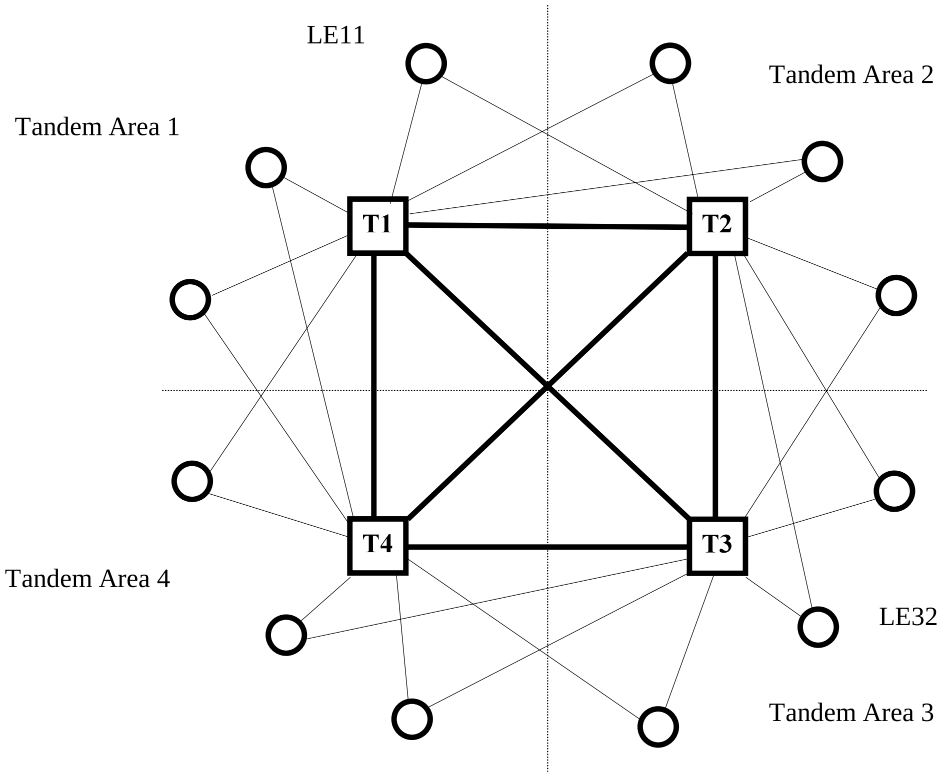


Figure 8.4: Double entrance network

Each local exchange (LE) is connected both to its own tandem exchange T and to another tandem. Tandem exchanges are fully inter-connected. High usage routes may be installed as well, but are not shown here.

The routing plan is influenced by, or is dependent on several other fundamental technical plans, as shown in Figure 8.5.

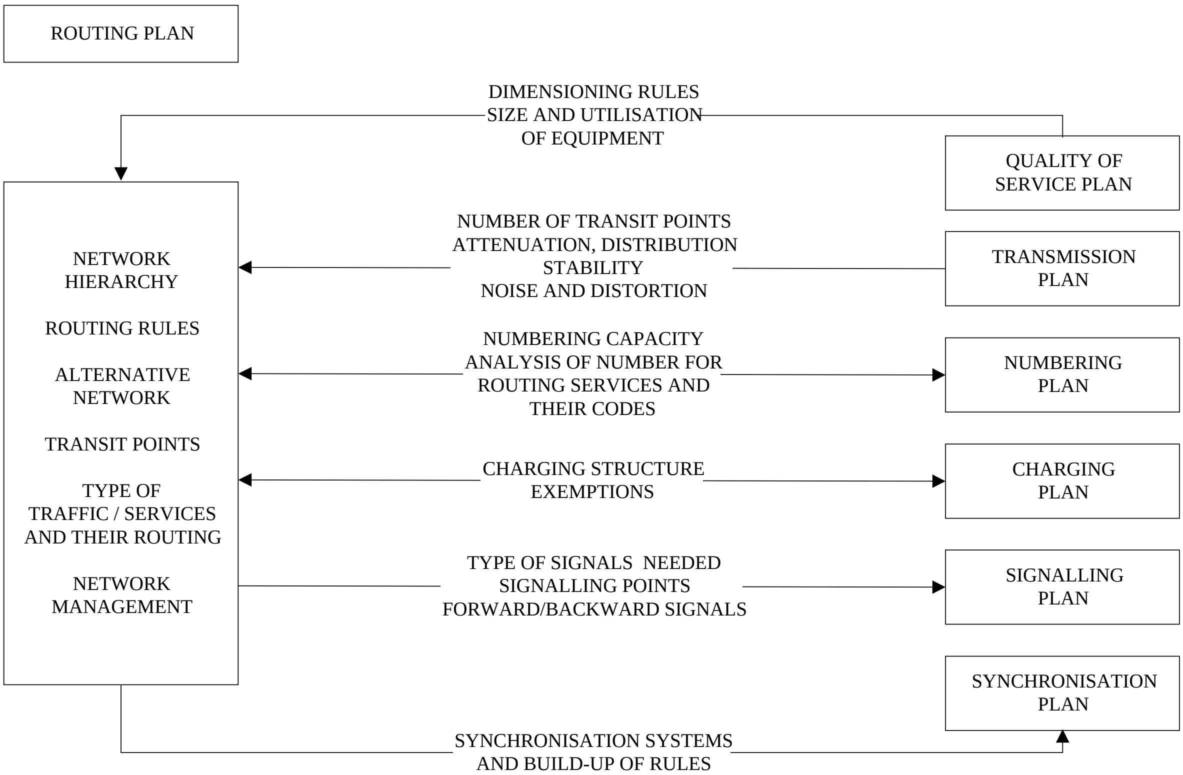


Figure 8.5 : Interdependence between the routing plan and other fundamental plans

8.2.2 Switching

As for all other components of the developing network, the switching function is changing, probably even more radically than other items. The development of switches has been one of continuously increasing their degree of sophistication by building in more and more intelligence in the exchanges in order to be able to provide more and more new services.

This trend is now completely broken, since a continuation would lead to unmanageable design, updating, compatibility and handling situations. The network intelligence will be more or less removed from the exchanges and concentrated at **Service Control Points**, (**SCP)**.

Instead of exchanges, reference will be made to **Network Nodes**, and the telecommunication network will be defined as a structure of **Nodes** and **Links**. The nodes are there to carry out **Network Functions**, and there will be a number of different **Node Types**, for example **Access Nodes**, **Transfer Nodes**, **Network Management Nodes**, etc. An access node, for instance, can be specialised for some kind of **Mobile** (cellular radio) access or ISDN access etc. A transfer node should also be specified according to functions, for example synchronous 4/1 DXC with PDH/SDH bridging, ATM switching and 64 kbit/s switching including ISDN in the node. Required capabilities of switches and related elements such as DXC should be specified, for example holding/switching times, signalling capabilities, overall traffic or call capacities, routing capabilities etc.

The final switching plan should give information such as:

a) general description of the major features, e.g. dates of 50% and of total conversion, exchange type policy, services which will be available;

b) implications for the customers;

c) overall cost, in total and per annum;

d) commercial implications;

e) political/prestige implications;

f) technical benefits;

g) list of switch types and related equipment, with their individual conversion policies;

h) any favouring of geographical areas, with reasons;

i) reasons for choosing this plan in preference to others;

j) a detailed list of all network nodes, their new equipment, and dates of implementation or conversion.

8.3 Signalling

8.3.1 General aspects

Signalling in telecommunication networks is applied for the establishment and supervision of connections in the network and can be generally divided into two categories:

- signalling between exchanges;
- signalling between exchanges and terminal equipment.

The degree of sophistication of a signalling system is closely related with the technology applied in the related switching or terminal equipment.

The old electromechanical step-by-step systems could only work with simple decimal pulse type signalling systems which only provide the essential signals for the establishment and supervision of the connections.

Register and common-control electromechanical systems could already use multifrequency interregister signalling systems such as the R2 system or derived versions of it, providing an extensive range of signals for category and identity of calling subscribers, information about network conditions during call establishment and conditions and categories of called subscribers.

It has to be recalled that, with the R2-interregister signalling, it is necessary to apply different line signalling systems depending on the transmission media, e.g. d.c. line signalling on two-wire analogue circuits, out-band high-level pulse-type or CCITT low-level continuous line signalling on analogue four-wire circuits, and CCITT T-16 line signalling on PCM-systems.

With the introduction of Stored-Programme-Controlled (SPC) switching systems, it became possible to introduce fast interexchange signalling via data links, and to apply common channel signalling systems instead of channel associated signalling used up to this time. This resulted in the creation of the first common-channel signalling system, CCITT No. 6, which is mainly used with analogue SPC-systems.

As far as digital SPC-systems are concerned, it took a long time before the latest common channel signalling system, CCITT No. 7, became available in a stable version, suitable for digital networks and for the introduction of ISDN.

The development of signalling systems between terminal equipment and the exchange has followed a similar pattern as for the interexchange signalling.

Initially, only loop-disconnect signalling was used for dial telephones, push-button telephones with d.c.-dialling, and for PABXs including Direct-Dialling-In (DDI) facilities.

With the standardisation of Dual-Tone-Multi-Frequency (DTMF) signalling for push-button telephone sets (CCITT Recommendation Q.23), the DTMF-signalling is also used with PABXs for Direct-Dialling-In. However, other multifrequency signalling systems can be used, such as the R2-system, depending on the facilities in both local exchanges and PABXs.

For ISDN-connections to subscribers and PABXs, D-channel signalling according to the CCITT Q.900-series of Recommendations will be applied, also known as DSS 1 (Digital Subscriber Signalling System), both for basic access (2B+D) to individual lines as well as for primary access (30B+D) for PABXs.

For signalling on analogue national circuits which incorporate domestic satellites, a pulsed version of CCITT R2 signalling has been used. CCITT System N° 5 can also be used in some cases> Digital circuits over satellite connections can use CCITT signalling system N° 7. If only analogue satellite circuits are available, the possibility of routing the signalling over terrestrial digital circuits must be investigated.

8.3.2 Inventory and policy regarding the application of interexchange signalling systems in national networks

An inventory of interexchange signalling systems in the national network could include the following categories of signalling systems:

a) in exceptional cases, some very old pulse-type signalling systems associated with old electromechanical exchanges;

b) very frequently, rather sophisticated channel associated signalling systems with multi-frequency interregister signalling, such as the R2-system or derived versions, not only between common-control or register controlled electromechanical exchanges, but also between analogue SPC and even between digital exchanges;

c) exceptionally, the (outdated) CCITT common-channel signalling system No. 6, between analogue SPC switching systems;

d) sometimes, early versions of Signalling System No. 7 with Telephone User Part (TUP) according to CCITT Yellow or Red Book Specifications (early 1980 standards);

e) possibly, recent and up-to-date versions of Signalling System No. 7 with ISDN User Part (ISUP) according to new CCITT standards (e.g. Recommendation Q.767) as well as regional standards in so far as they exist (for example Europe and Africa).

Regarding the application of old signalling systems under a), the number of interfaces with newer switching equipment should be limited as much as possible; the possibility of signalling converters might be considered as a possible means to avoid the introduction of special interface equipment in new (digital) exchanges.

The question of the continued use of systems with multifrequency interregister signalling as mentioned under b), together with the introduction of an up-to-date version of Signalling System No. 7 as indicated under e) is a complex problem.

On the one hand, an Administration may argue that in view of the low development and the need for basic services, it will not need ISDN and therefore the introduction of a new and complicated signalling system with its staffing requirements can be postponed for the time being.

On the other hand, one may argue that in view of the extension of the network, which will obviously be important compared to the existing equipment, it might be an advantage to shift immediately to Signalling System No. 7 to be prepared for the future and its new service requirements, and to avoid investments in channel associated signalling systems which will not be able to meet new service requirements.

However, it should be kept in mind that for the introduction of Signalling System No. 7, it will be necessary to have a synchronised digital network and the Administrations should have an adequate number of qualified staff to deal with Signalling System No. 7.

If the introduction of Signalling System No. 7 is adopted, then it will be necessary to define the signalling network with Signalling Points (SPs) and possibly Signalling Transfer Points (STPs) as well as other procedures relevant to the network.

In the evolving network, it is possible that more services will be offered to analogue subscribers and it may be necessary to increase the number of signals used in the conventional interregister signalling. This may be easy to implement because normally a large number of signals available in an interregister signalling system such as R2 are not being used (for example the transmission of the calling subscriber number).

In these cases, the interworking arrangements between national signalling systems as well as between national and international signalling systems shall be such that the interchange of the information from forward and backward signals from one signalling system to another signalling system shall be as complete as possible.

As far as point d) is concerned, the application of early versions of Signalling System No. 7, they should be modified at a certain stage to the final version of Signalling System No. 7 adopted in the network.

For certain signalling systems, it will be necessary to make special arrangements on transmission circuits with longer propagation delays.

For example, with Signalling System No. 7, the basic error correction method will be applied for signalling links where the one-way propagation time is lower than 15 ms, whereas the preventive cyclic retransmission method applies for intercontinental signalling links with a one-way propagation time greater than or equal to 15 ms and for all links established via satellite.

For the Signalling System R2, the introduction of semi-compelled or non-compelled working of the interregister signalling has to be considered for signalling via satellite links in order to decrease the postdialling delay on this type of connections.

8.3.3 Signalling on international circuits

The signalling systems to be considered for international working are the following systems standardised by CCITT:

- CCITT Signalling System No. 5, for satellite and other intercontinental connections;
- CCITT Signalling Systems R1 and R2, for regional traffic, the R1 system being applied in North America, the R2 system in the other world regions;
- CCITT Signalling System No. 7, which may be used when digital switching and transmission equipment is involved, both for regional and intercontinental applications.

CCITT Signalling System No. 6 is used on a limited scale only, on some submarine cable systems across the Atlantic Ocean and the Pacific Ocean. With the development of Signalling System No. 7, System No. 6 became obsolete, and its use should be avoided.

When Digital Circuit Multiplication Equipment (DCME) is applied in relation with satellite links, then the international exchange concerned should be capable of providing the special signalling arrangements with this equipment as included in CCITT Recommendation Q.50.

For cross-border traffic with neighbouring countries, special (national) signalling schemes may be applied, to be agreed with the partners concerned. Sometimes there are regional agreements concerning the type of signalling to be applied for these types of connections (for example PANAFTEL R2 in Africa).

In this case, the selection procedures are based on normal dialling procedures for national traffic, call charging by normal long-distance rates.

8.3.4 Signalling between local exchanges and terminal equipment

A review has to be made of the existing signalling arrangements between local exchanges and terminal equipment such as telephone sets, fax machines and PABXs. Limitations in loop resistance, which would unnecessarily limit the distance between terminal equipment and the local exchange, should be avoided.

For loop-disconnect systems with dial pulsing or DTMF selection, the issue of metering signals might be considered. If this facility is not yet available, but would be desirable in view of certain charging facilities, then such a system could be introduced, preferably a low-level system using pulses of 12 or 16 kHz.

If the old high-level 50 Hz system is used, consideration should be given to the replacement of this system by the 12 or 16 kHz system.

Regarding Direct-Dialling-In to PABXs, it would be useful to specify the signalling systems available for this facility (R2-signalling, DTMF-signalling or a decimal pulsing system, either one or more possibilities).

Concerning the DSS 1 signalling system for the ISDN basic access via copper pairs and for subscriber loops covering longer ranges, it is recommended to use the Adaptive Hybrid Technique (echo cancellation).

As far as the modulation format is concerned, the actual trend is to adopt the system based on a line code 2B1Q, a standard widely used, and for which equipment can be obtained from many sources.

(In the case of signalling between an exchange and a RSU a sub-set of CCS N° 7 is used. It should be remembered that systems of different manufacturers are not always compatible).

8.3.5 Signalling in special public networks

In some special networks, it will be necessary to adopt special signalling procedures, the most common example being mobile cellular radio networks.

In such networks, it is necessary to know the location (cell) of the mobile subscribers, and if such a network consists of a number of switching nodes, then it will be necessary to exchange this information between these switching nodes by means of a suitable signalling arrangement.

Signalling System No. 7 is capable of providing this facility by means of its Mobile Application Part (MAP), and is therefore the most suitable signalling system in such a network.

In such a case, a signalling network including network structure and SP-numbering should also be defined.

The CCITT / ITU-T Q - series of Recommendations provide for interworking between signalling systems.

8.3.5 References

1. CCITT: Economical and Technical Aspects of the Choice of Telephone Switching Systems, ITU, 1981.

2. CCITT Blue Book (1989) Volume VI:

   a. Fascicle 1: General Recommendations on Telephone Switching and Signalling;

   b. Fascicle 2: Specifications of Signalling Systems No. 4 and No. 5;

   c. Fascicle 4: Specifications of Signalling Systems R1 and R2;

   d. Fascicle 7, 8 and 9: Specifications of Signalling System No. 7;

   e. Fascicle 10 and 11: Digital Subscriber Signalling System No. 1 (DSS 1);

   f. Fascicle 12 and 13: Public Land Mobile Network.

3. Post-Blue Book recommendations concerning Signalling System No. 7 and DSS 1:

   - Recommendation Q.767: Application of the ISUP of Signalling System No. 7 for international ISDN interconnections;

   - Recommendations Q.951 and Q.952: Stage 3 service description for number identification and call completion supplementary services, using DSS 1.

4. CCITT: Guidelines for Implementing a Signalling System No. 7 Network (ITU, 1991).

5. Guidelines and Specifications for the Implementation of Signalling System No. 7 in the PANAFTEL Network for National and Regional Applications (ITU/UNDP, 1992).

8.4 Transmission plan

In many cases, the transmission plans still used by Administrations have been found to be both incomplete and out of date, in view of the pace of digitalisation and of the modifications made to the relevant CCITT Recommendations in recent years. These modifications, mainly concerning the adoption of loudness ratings (LR) in place of the previously used reference equivalents (RE and CRE), are reviewed in this section together with the implications of digitalisation - in particular the increasing use of digital local exchanges and remote subscriber units (RSU).

8.4.1 Objectives of a transmission plan

The principal objectives of a transmission plan are:

- to define the losses and other impairments which can be tolerated over each part of any connection to ensure satisfactory communication between any two subscribers irrespective of their location on the national or international network;

- to specify the technical means by which the above objectives can be achieved.

The first of the above is the subject of this section while the second, as far as transmission media are concerned, is considered in paragraph 9.4. Local network planning and the routing and switching plan are also closely related; these are described in paragraphs 9.2 and 8.2 respectively.

8.4.2 CCITT Recommendations

The latest CCITT Recommendations concerning transmission planning are included in Volume III - Fascicle III.1 and Volume V of the Blue Books which contain the relevant Recommendations applicable after the IXth Plenary Assembly held at Melbourne in November 1988. As already mentioned these Recommendations have been modified and it is not possible to detail all these modifications in the present document. It is therefore recommended that the above volumes of the Blue Books, together with the now separately published ITU-T Recommendations which are applicable, are made available to the staff involved in transmission planning and used as references for the interpretation of the information presented here.

In particular the following Recommendations should be carefully studied:

- G.101: The Transmission Plan, which includes some essential definitions;

- G.102: Transmission performance objectives and Recommendations;

- G.103: Hypothetical reference connections;

- G.111: Loudness Ratings (LRs) in an international connection, Annex D of which includes an explanation of the reasons for using LRs;

- G.113: Transmission impairments, which includes limits applicable to digital connections;

- G.121: Loudness Ratings (LRs) of national systems, Annex C of which includes loss assignments at the 4-w audio points of digital systems;

- G.131: Stability and echo, mainly applicable to long international connections;

- G.142: Transmission characteristics of exchanges, which includes requirements for digital exchanges;

- P.11: Effect of transmission impairments, dealing with the subjective effect of these impairments on subscriber opinion and also containing useful background information on the use of LRs;

- P.76: Determination of loudness ratings; fundamental principles, containing useful background information on LRs.

8.4.3 Reference equivalents and loudness ratings

Until comparatively recently, the transmission quality of telephone connections and their component parts, as far as loss was concerned, was expressed in terms of "Reference Equivalents" (RE) in the sending and receiving directions - the "reference" being the CCITT standard known as "NOSFER" against which the real connection is compared.

The reference equivalent gives the difference in dB, for the send and receive directions, between the system to be examined and the reference. It therefore provides a measure of the transmission loss, from mouth to ear, of a speech path.

Reference equivalents were first defined in the CCITT Orange Books and were redefined, as "corrected" reference equivalents (CRE) in the Yellow Books and in the Red Books, in which the concept of loudness rating (LR) was introduced. Conversion formulae and tables were included in these Recommendations.

The reference equivalent method for defining transmission quality, which is based on the criterion of loudness of speech emitted by the talker and perceived by the listener, is defined in CCITT Recommendations P.42 and P.72 (Red Books) - which are not superseded. The method for determining loudness ratings of local telephone circuits is based on similar fundamental principles but incorporates modifications which provide more flexibility. In particular, reference equivalents cannot be added algebraically - often discrepancies of up to 3 dB can occur - while the LRs of the different segments of a connection can be simply added together. Also the replication accuracy of reference equivalents, as measured in the laboratory, is poor.

As is the case for reference equivalents, loudness ratings, expressed in the dimensions of "loss", are defined by the amount of loss inserted in a reference system to secure equality of perceived loudness to that obtained over the speech path being measured.

The nominal overall loudness rating (OLR) of a complete international connection is the sum of:

- the nominal send loudness rating (SLR) of the national sending system;
- the nominal circuit loudness rating (CLR) of the international chain;
- the nominal receive loudness rating (RLR) of the national receiving system.

This elementary division is shown in Figure 8.6 below.

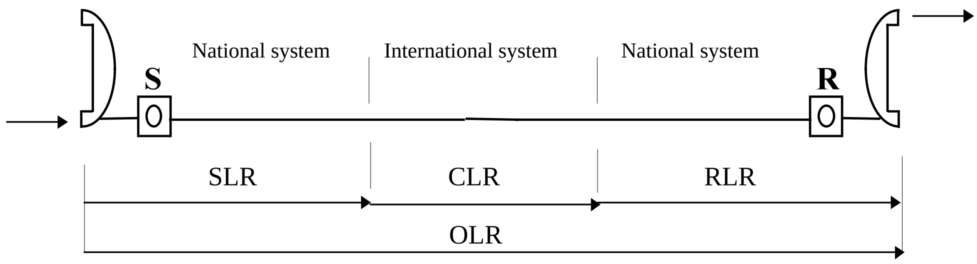


Figure 8.6: Designation of LRs in an international connection

As shown in the following table, the nominal (or optimum) value of OLR is in the order of 10 dB for practical connections. The table, reproduced from CCITT Recommendation G.111 (Blue Book), also indicates the changes made in the recommended traffic-weighted mean values of loudness ratings since 1984. These are given as objectives, with values referred to both the 0 dBr and the virtual analogue switching point (VASP) of the international exchange which, for the digital case, are shown in Figure 8.7.

TABLE 8.4.1

Values (dB) of sending, receiving, circuit and overall loudness rating cited in Recommendations G.111 and G.121

	Recommended in 1984			Recommended in 1988					
	SLR	RLR	OLR	SLR		RLR		CLR	OLR
	VASP	VASP		0 dBr	VASP	0 dBr	VASP		
Optimum value			» 5						» 10
Traffic-weighted mean values:									
long-term objective (minimum)	6.5	-2.5	8	7	10.5	1	- 3	(Note 1)	8
long-term objective (maximum)	8	-1	11	9	12.5	3	- 1	(Note 1)	12
short-term objective (maximum)	14	2.5	20.5	15	18.5	6	2	(Note 1)	21
Maximum values for an average-sized country	20	9		16.5	20	13	9	*n* x 0.5 (Note 2)	
Minimum for sending	2			- 1.5	2				

Note 1 - *CLR* = 0 for a digital international circuit, 0.5 dB for an analogue one. The average number of international circuits is about 1.

Note 2 - *n* is the number of analogue international circuits.

Note 3 - The VASPs are defined in Recommendation G.1101.

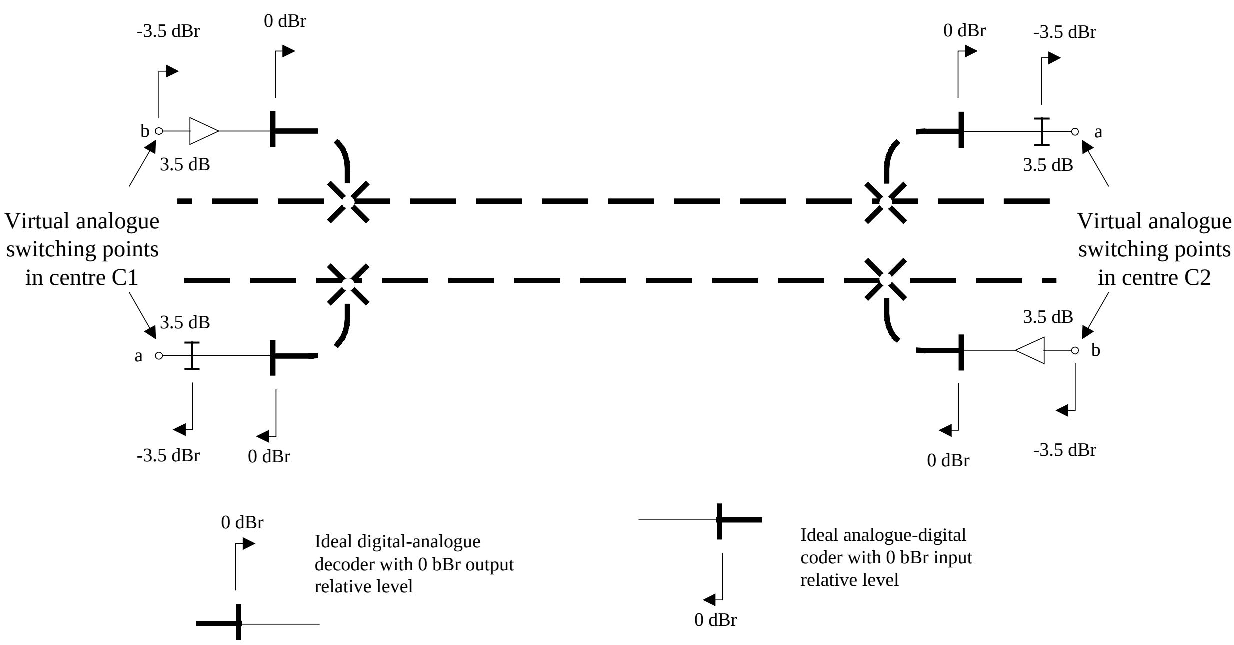


Figure 8.7: Virtual analogue switching points

8.4.4 Maximum values of SLR and RLR for national systems

The nominal maximum values of SLR and RLR, in each direction of transmission, for the national segments of a connection are recommended in CCITT G.121 as shown in Table 8.4.2.

TABLE 8.4.2

Nominal maximum LRs recommended for national systems

Country size [a]	No. of nat. circuits in the 4-w chain [b]	0 dBr point		VASP	
		SLR	RLR	SLR	RLR
Average	Up to 3	16.5	13	20	9
Large	4	17	13.5	20.5	9.5
Large	5	17.5	14	21	10

As already indicated, although LRs can be defined at any interface of the network, the SLR and RLR of the national segments are normally specified at the international exchange interface - either at the 0 dBr test point or the virtual analogue switching point (VASP) as shown in the above table. In CCITT Recommendations G.111 and G.121, the SLRs and RLRs of national systems are referred to a 0 dBr test point at the international exchange.

8.4.5 Minimum value of SLR

In order to prevent overloading of the international transmission systems, a minimum value of SLR has to be respected. This value is provisionally recommended to be -1.5 dB with respect to the 0 dBr test point or 2 dB when referred to the VASP. It should be noted that a low SLR can result from the use of unregulated telephone sets. In this context, the speech power applied to international circuits by operators sets must be controlled. It is recommended therefore that regulated telephone sets should be exclusively used.. (In practice replacement sets should be of the regulated type).

8.4.6 Sidetone

Among the other transmission impairments to be considered is an inadequate sidetone performance of the various equipment combinations used to establish a connection. This applies in particular to connections approaching the upper limits for loudness rating and/or noise performance. For 2-wire telephone sets the sidetone performance depends on the set sensitivity and line impedance variations and, ideally, the set should be designed with an adaptive sidetone balancing function to widen the acceptable range of line impedances. This ideal solution, however, is far too costly to incorporate in standard telephone sets and the strategy usually adopted is to control the network impedances.

The sidetone performance is expressed in terms of:

- the talker's sidetone masking rating (STMR);
- the listener's sidetone rating (LSTR),

which are explained in detail in CCITT Recommendations G.111, G.121, P.76 and P.79.

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[a] See Recommendation G.101, § 2.2.

[b] Analogue or mixed analogue/digital

Note - When comparing these maximum values of LRs with LRs determined for existing networks some discrepancies may be found. If the actual LRs are greater by 2 or even 3 dB this is no cause for concern. On the other hand, if a margin of 2 or 3 dB seems to appear, the permissible attenuation for subscriber lines should not automatically be increased. The first step should instead be to use the margin to improve the traffic-weighted mean values referred to in § 1.2.

For planning purposes, the preferred range of STMR is recommended to be 7 - 12 dB and the minimum value of LSTR should be 13 dB (in the ideal case).

8.4.7 Other transmission impairments

In addition to those mentioned above, other impairments which can affect the transmission quality of a connection are briefly reviewed in the following sections. The acceptable limits for these impairments are given in the CCITT G.100-Series of Recommendations and their effects on subscriber opinion are considered in CCITT Recommendation P.11.

8.4.8 Attenuation and group-delay distortion

Ideally this should be uniform over the voice frequency band. In practice, this is never the case and limits have to be specified with respect to the centre frequency (800 Hz). These well-known limits are given in CCITT Recommendations G.132 (Figure 8.1) and G.133 (Table 1). As far as digital systems are concerned the problem of this type of distortion can be minimised by having as few analogue to digital conversions as possible in the network (CCITT Recommendation G.113).

8.4.9 Stability and echo

Referring to Figure 8.8, the transmission loss introduced between the points a and b by the national system (loss a-b) contributes to the margin available to the international connection against oscillation during the setting-up and clearing down of the connection. It also provides an essential margin of stability during a communication and protection against echoes. In addition, it can be seen that echoes circulating in any 4-wire loop in the national or international chains can give rise to listener echo which can affect voice-band analogue data transmission.

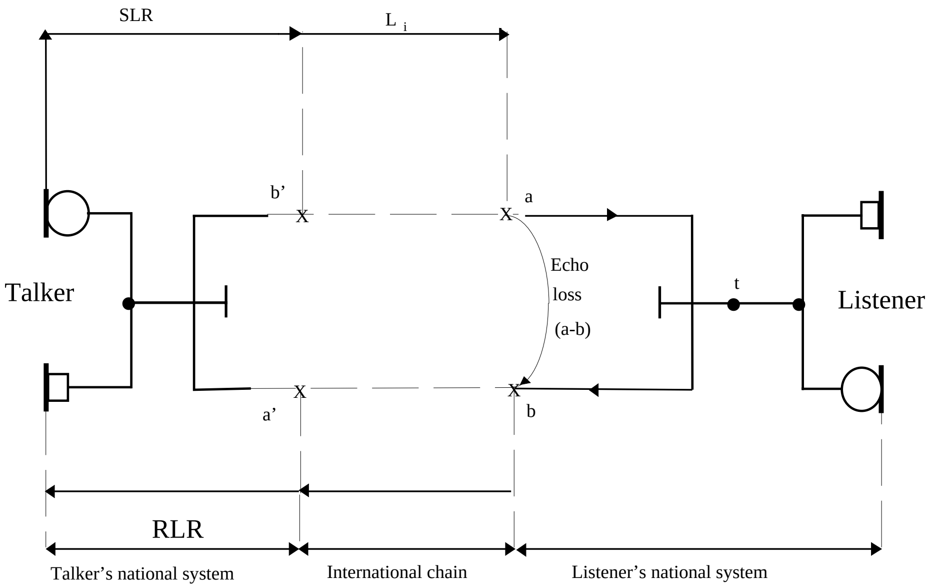


Figure 8.8: Echo paths

The objectives for the loss a-b are given in CCITT Recommendation G.122, Annex B of this Recommendation including a definition of the overall loudness rating of the echo path. From Figure 8.8 this is seen to be:

$$SLR + RLR + 2L_i + (\text{loss a-b})$$

This subject is considered further in CCITT Recommendation G.131, in which some practical rules are given and the effects of delay assessed.

8.4.10 Crosstalk

This consists of interference from other circuits, either at the far-end or the near-end. Poor crosstalk ratios are usually due to badly laid-out transmission systems, in particular multi-pair cables, and faulty exchange equipment. It is advisable to establish rather stringent limits for crosstalk ratios, for example 65 dB as (until recently) it is usually masked by other noise sources. With the improved quality of digital circuits, poor crosstalk ratio will be more evident. It is of course one of the crucial factors to be considered when selecting pairs for PCM transmission. Further information is given in CCITT Recommendation G.151 for particular cases.

8.4.11 Noise, and quantising distortion

The concept of noise power allocation which is directly proportional to the circuit length is used for analogue networks, the allocations being given in CCITT Recommendations G.123 and G.222 for example, G.123 also including the allocations for switching equipment and noise induced by power lines.

This concept is not strictly applicable in the case of a digital network; for example circuits designed to meet the objectives of the CCITT G.700-Series Recommendations will exhibit a noise performance which is substantially independent of the circuit length. In general, the quality of a connection established over digital facilities is determined by the bit-error-rate (BER) impairments due to noise, quantisation distortion, inter-symbol interference, etc. in the various segments causing corresponding degradations of the BER.

Quantisation distortion is introduced by the PCM coding process so that the number of codecs in any connection should be minimised. It should be remembered that codecs are also used in the digital switching equipment so that the contribution from this source can be of the same order as that from the digital transmission links. Thus the number of codecs, and therefore the number of digital switches, in a worst-case international connection has to be limited for example to 14 according to CCITT Recommendation G.142. For a national network the number of codecs should be limited to 5 (See CCITT Recommendation G.113)

In addition to the BER degradation due to analogue/digital conversions, etc. already mentioned, the BER can also be affected by the frame slips caused by inadequate synchronisation of the exchange "clocks" i.e. by the outgoing and incoming bit rates of the digital signal between two exchanges not being identical. The effect of poor synchronisation is more pronounced on data and facsimile circuits, the subjective effect of a slip on an audio circuit being negligible. As the digital networks will inevitably be used for services other than telephony the permissible slip-rates, and the methods of synchronisation to be used, have to be defined. This subject is outside the scope of a transmission plan and is covered in the separate network synchronisation plan.

8.4.12 Jitter

While jitter has little influence on the telephony service, it could interfere with digital facsimile and video services. Jitter, which is defined as the short-term variations of the significant instants of a digital signal from their ideal positions in time, is covered in CCITT Recommendation G.823.

8.4.13 Loss assignment on the digital network

The expansion of digital switching and transmission over the network will result in significant improvements to the transmission quality of those connections now made using analogue facilities, in particular to the effective losses encountered. The introduction of digital local exchanges, and junctions, is merely following the principle of extending 4-wire facilities to the local level, i.e. as near as possible to the subscribers, thus eliminating the major, and variable, 2-wire losses.

The 7 dB loss normally assigned to an analogue 4-wire chain is provided by the insertion losses of the hybrids at the circuit extremities. A similar value can be assigned to the inherently 4-wire digital chain to provide the loss required to maintain stability and control the echo effects. In this case the loss can be represented by a fixed pad (or pads) at the analogue points of the circuit extremities. The value of 7 dB (total) is now commonly used by many administrations as shown in Table C-1 of CCITT Recommendation G.121. This concept, extended to the local level, is shown in Figure 8.9.

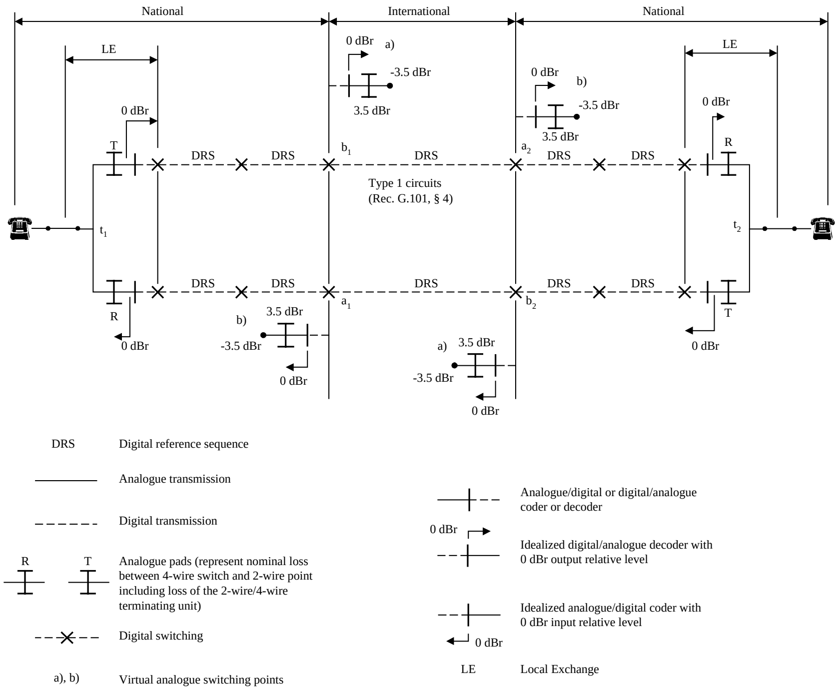


Figure 8.9: Extension of 4-w switching to LE

The principal advantage of this arrangement is of course that <u>all</u> connections made over the national network, whether local or long-distance, would have practically the same quality, only variations in the losses of the subscriber lines being present. Thus no special networks, formerly used to attempt to equalise the connection losses, are required.

As far as the OLR is concerned, there could obviously be an immediate improvement, to approach the CCITT short-term objective, even with the existing subscriber networks. The OLR in this case is determined only by that of the local telephone systems.

8.4.14 Summary of transmission planning objectives

The following is a résumé of the planning objectives which should be incorporated in a national transmission plan and gives a range of values which are considered appropriate for a typical network.

It is assumed that this typical network is at a fairly advanced, but not complete, stage of digitalisation and is located in a country of average size. In addition, it is assumed that:

- the national telephone network will be fully automatic;
- all switching will be carried out on a 4-wire basis;
- all the inter-exchange transmission, including junction networks, is 4-wire;
- all the above are established by means of high-velocity transmission media (radio-relay, optical fibre, etc.);
- the maximum length of any national connection does not exceed about 1,250 km.

a) <u>Loudness ratings</u>

With the above assumptions, the short-term objectives recommended by the CCITT can be met for the majority of connections established over the network.

The following values for loudness ratings should be used as planning objectives:

i) Maximum values

- with reference to the 0 dBr test point at the ITSC:

SLR : 16.5 dB
RLR : 13.0 dB

- with reference to virtual analogue switching point:

SLR : 20.0 dB
RLR : 9.0 dB

ii) Minimum value of SLR

- with reference to the 0 dBr test point at the ITSC:

SLR : -1.5 dB

- with reference to virtual analogue switching point:

SLR : 2.0 dB

iii) Traffic-weighted mean values of LR.

These values are required to ensure that the majority of subscriber are provided with satisfactory transmission, which would not be the case if the maximum values specified above were consistently applicable.

The following mean values, corresponding to those recommended by the CCITT as short-term planning objectives, should be used:

- with reference to the 0 dBr test point at the ITSC:

SLR : 7 to 15 dB
RLR : 1 to 6 dB

- with reference to the virtual analogue switching point:

SLR : 10.5 to 18.5 dB
RLR : -3.0 to 2.0 dB

b) <u>Sidetone masking rating and listener's sidetone rating</u>

Reference should be made to CCITT Recommendations G.111, G.121, P.76 and P.79.

The planning values to be used as objectives should be as follows:

- sidetone masking rating (STMR): 7--12 dB

- listener's sidetone rating (LSTR): 10-15 dB

In practice, the STMR depends on the telephone set values of SLR and RLR, and the sidetone balance impedance. If the telephone set SLR and RLR are known the STMR can be calculated using the algorithm given in CCITT Recommendation G.111, Annex 1.

In the case of the LSTR the following relationship applies in practice:

LSTR = STMR + D

where D has been found to be 1.5 to 4.0 dB.

c) Loss allocation to digital local exchange

Reference should be made to CCITT Recommendation G.121, Annex C.

In order to meet the stability requirements of CCITT Recommendation G.122 the value of 7 dB is recommended for the loss of the combined "R" and "T" pads to be provided at the analogue points of digital local exchanges connected to 2-wire local networks.

The 4-wire digital switching function itself should present zero transmission loss, i.e. if a 0 dBm0 sinusoidal test signal is introduced at the analogue terminals of an ideal coder connected to the input of a digital switch, a digital reference sequence (DRS) should be transmitted unaltered through the switch and produce a 0 dBm0 sinusoidal signal at the analogue terminals of a decoder connected to the output of the digital switch (see also CCITT Recommendation G.142).

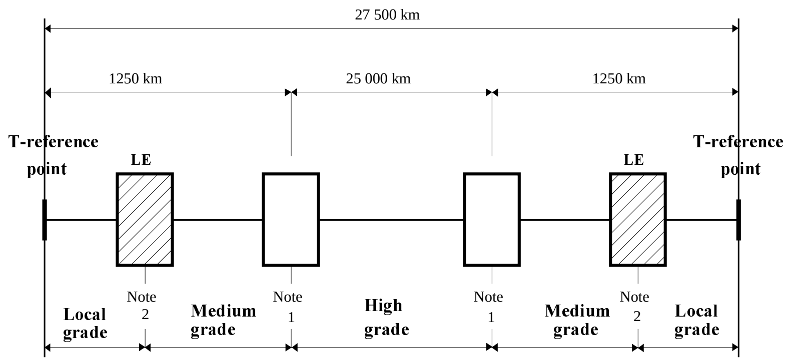
d) Circuit noise and bit error ratios (BER)

Contributions to circuit noise are principally from long or medium connections using analogue transmission facilities over which the noise power is typically proportional to the circuit length (L km). As a general objective, the total noise power from this source should not exceed the value 3L pWp0.

The effect of random noise (and of other impairments) in a digital connection is a degradation of the bit error ratio (BER). Error performance objectives are specified by both the CCITT and the CCIR for digital connections forming part of an ISDN and are given in the following Recommendations:

- CCITT Recommendation G.821
- CCITT Recommendation G.921
- CCIR Recommendation 594
- CCIR Recommendation 634
- CCIR Recommendation 696
- CCIR Recommendation 697.

The error performance objectives are specified for several types of hypothetical and real circuits and are classified according to the diagram below (Figure 1/G.821 of CCITT Recommendation G.821):



Note 1 - It is not possible to provide a definition of the location of the boundary between the medium and high grade portions of the HRX. Note 4 to table 2/G.821 provides further clarification of this point.

Note 2 - LE denotes the local exchange or equivalent point.

Figure 1/G.821:Circuit quality demarcation of longest HRX

The performance objective is stated in terms of error performance parameters each of which is defined as follows (according to CCITT Recommendation G.821):

"The percentage of averaging periods each of time interval T_o during which the BER exceeds a threshold value. The percentage is assessed over a much longer time interval T_l" (for example any one month)

The following BERs and intervals are used in the statement of objectives:

- a BER of less than 1.10^{-6} for T_o = 1 minute
- a BER of less than 1.10^{-3} for T_o = 1 second
- zero errors for T_o = 1 second (equivalent to the concept of error free seconds EFS)

These categories equate to those in the following table (Table 1/G.821 of Recommendation G.821).

Table 1/G.821

Error performance objectives for international ISDN connections

Performance classification	Objective (Notes 3, 5)
(a) (Degraded minutes) (Notes 1, 2)	Fewer than 10% of one-minute intervals to have a bit error ratio worse than $1 \cdot 10^{-6}$ (Note 4)
(b) (Severely errored seconds) (Note 1)	Fewer than 0.2% of one-second intervals to have a bit error ratio worse than $1 \cdot 10^{-3}$
(c) (Errored seconds) (Note 1)	Fewer than 8% of one-second intervals to have any errors (equivalent to 92% error-free seconds)

Note 1 - The terms "degraded minutes", "severely errored seconds" and "errored seconds" are used as a convenient and concise performance objective "identifier". Their usage is not intended to imply the acceptability, or otherwise, of this level of performance.

Note 2 - The one-minute intervals mentioned in Table 1/G.821 and in the notes (i.e. the periods for M > 4 in Annex B) are derived by removing unavailable time end severely errored seconds from the total time and then consecutively grouping the remaining seconds into blocks of 60. The basic one-second intervals are derived from a fixed period.

Note 3 - The time interval T_L , over which the percentages are to be assessed has not been specified since the period may depend upon the application. A period of the order of any one month is suggested as a reference.

Note 4 - For practical reasons, at 64 kbit/s, a minute containing four errors (equivalent to an error ratio of 1.04×10^{-6}) is not considered degraded. However, this does not imply relaxation of the error ratio objective of 1×10^{-6}.

Note 5 - Annex B illustrates how the overall performance should be assessed.

The following tables, with explanatory notes, are extracted from CCITT Recommendations G.821 and G.921 and give the distribution of error performance objectives over the three circuit classifications mentioned above (local grade, medium grade and high grade) and over section quality classifications on shorter connections. The latter take account of the use of digital radio-relay systems.

Table 2/G.821

Allocation of the degraded minutes and errored seconds objectives for the three circuit classifications

Circuit classification	Allocation of the degraded minutes and errored seconds objectives given in Table 1/G.821
Local grade (2 ends)	15% block allowance to each end (Notes 1, 4 and 5)
Medium grade (2 ends)	15% block allowance to each end (Notes 2, 4 and 5)
High grade	40% (equivalent to conceptual quality of 0.0016% per km for 25 000 km, but see Note to § 3.1) (Notes 3, 6 and 7)

Note 1 - The local grade apportionment is considered to be a block allowance, i.e. an allowance to that part of the connection regardless of length.

Note 2 - The medium grade apportionment is considered to be a block allowance, i.e. an allowance to that part of the connection regardless of length. The actual length covered by the medium grade part of the connection will vary considerably from one country to another. Transmission systems in this classification exhibit a variation in quality falling between the other classifications.

Note 3 - The high grade apportionment is divided on the basis of length resulting in a conceptual per kilometre allocation which can be used to derive a block allowance for a defined network model (e.g. Hypothetical Reference Digital Link). For practical planning purposes of links in network models, link allowances based on the number of 280 km sections nominally 280 km (as specified in Table 2/G.921) can be used in place of the per kilometre allocation specified in this Recommendation. For longer sections which are not an exact integer multiple of 280 km, the next higher integer multiple is used.

Note 4 - The local grade and medium grade portions are permitted to cover up the first 1250 km of the circuit from the T-reference point (see Figure 1/G.821) extending into the network. For example, in large countries this portion of the circuit may only reach the Primary Centre whilst in small countries it may go as far as the Secondary Centre, Tertiary Centre or the International Switching Centre (see Figure 1/G.821).

Note 5 - Administrations may allocate the block allowances for the local and medium grade portions of the connection as necessary within the total allowance of 30% for any one end of the connection. This philosophy also applies to the objectives given for local and medium grades in Table 3/G.821.

Note 6 - Based on the understanding that satellite error performance is largely independent of distance, a block allowance of 20% of the permitted degraded minutes and errored second objectives is allocated to a single satellite HRDP employed in the high-grade portion of the HRX.

Note 7 - If the high grade portion of a connection includes a satellite system and the remaining distance included in this category exceeds 12 500 km or if the high-grade portion of a non-satellite connection exceeds 25 000 km, then the objectives of this Recommendation may be exceeded. The occurrence of such connections is thought to be relatively rare and studies are continuing in order to investigate this. The concept of satellite equivalent distance (the length of an equivalent terrestrial path) is useful in this respect and it has been noted that a value in the range 10 000 to 13 000 km might be expected.

Note 8 - For subscriber premises installation, between the T-reference point and terminal equipment, no specific requirements are given. However careful attention should be paid to the choice of the subscriber equipment since the overall performance of the connection depends heavily, not only on the network performance, but also on the quality of the terminal installation.

Apportionment strategy for severely errored seconds

The total allocation of 0.2% severely errored seconds is subdivided into each circuit classification (i.e. local, medium, high grades) in the following manner:

a) 0.1% is divided between the three circuit classifications in the same proportions as adopted for the other two objectives. This results in the allocation as shown in Table 3/G.821.

TABLE 3/G.821

Allocation of severely errored seconds

Circuit classification	Allocation of severely errored seconds objective
Local grade	0.015% block allowance to each end (note 5 to Table 2/G.821)
Medium grade	0.015% block allowance to each end (note 5 to Table 2/G.821)
High grade	0.04% (Notes 1,2)

Note 1 - For transmission systems covered by the high grade classification each 2500 km portion may contribute not more than 0.004%.

Note 2 - For a satellite HRDP operating in the high grade portion there is a block allowance of 0.02% severely errored seconds (see also Note 6 to Table 2/G.821).

b) The remaining 0.1% is a block allowance to the medium and high grade classifications to accommodate the occurrence of adverse network conditions occasionally experienced (intended to mean the worst month of the year) on transmission systems. Because of the statistical nature of the occurrence of worst month effects in a world-wide connection, it is considered that the following allowances are consistent with the total 0.1% figure:

- 0.05% to a 2500 km HRDP for radio relay systems which can be used in the high grade and the medium grade portion of the connection;

- 0.01% to a satellite HRDP (the CCIR are continuing studies on severely errored seconds performance for satellites systems and this value may eventually need to be increased).

Table 2/G.921

Digital section quality classifications for error performance

Section quality classification	HRDS length (km) (see Figure 4/G.801) (Note 2)	Allocation (Notes 3,4)	To be used in circuit classification (see Figure 1/G.821) (Notes 5 and 6)
1	280	0.45%	High grade
2	280	2%	Medium grade
3	50	2%	Medium grade
4	50	5%	Medium grade

Note 1 - There is no intention to confine any quality classification to any specific bit rate. The possibility of introducing additional options (for instance concerning length) requires further study.

Note 2 - The indicated values of length are those identified in Recommendation G.801. They should be understood to correspond to maximum lengths of real digital sections. If a real digital section is shorter, there will be no reduction of the bit error allocation (i.e. percentage value in the third column). This takes into account that :

- in many line systems (especially on copper wire pairs) most bit errors occur at the ends of the system;
- in the interest of economy, short-haul systems may be designed with greater per-kilometre error ratio than long-haul systems.

If a real digital section is longer (e.g. 450 km), its overall allocation should correspond to that of an integer number of HRDSs (of the same quality classification) the combined lengths of which are at least as long as the real section length (e.g. 2 x 280 km).

Note 3 - The values in this column are percentages of the overall degradation (at 64 kbit/s) specified in Recommendation G.821; i.e. of the 8% errored seconds, of the 10% degraded minutes and of the 0.1% severely errored seconds which are allocated according to the same rules as the two other parameters.

Note 4 - To obtain 64 kbit/s error performance data from error measurement at primary bit rates and above, the method described in Recommendation G.821, Annex D, should be used.

Note 5 - May also be used within a lower grade portion of the connection as defined per Figure 1/G.821.

Note 6 - To take account of adverse propagation conditions on radio systems as detailed in Recommendation G.821, an additional percentage of 0.05% of severely errored seconds has been allocated to a 2500 km radio-relay HRDP for systems operating in the high and medium grade quality part of the HRX. This corresponds for a 280 km section to a value of 0.0055% to be added to section quality classification 1 and 2 allocation when applied to severely errored seconds.
This would result in an additional allowance of 0.025% of severely errored seconds available for the medium grade part of the connection if it is realised entirely with class 1 radio sections. Where the medium grade portion of the network is realised with a mixture of different classifications, part of this additional allowance may be allocated to classes 3 and 4 at the discretion of Administrations.
To be consistent with the statistical assumptions made in G.821 § 3.3 b) regarding the number of radio sections in the HRX, and the occurrence of worst month effects it may be necessary to take into account the probability of worst month effects occurring simultaneously for all radio sections in a connection. A statistical model to be used for network planning and performance evaluation to assess the consistency of a given connection to the overall objective of G.821 is under study.

e) Attenuation distortion

The limits for a world-wide chain of 4-wire circuits given in CCITT Recommendation G.132, and shown in the figure below, should be respected.

These limits can easily be met over any national connection provided that an excessive number of codecs etc. is avoided.

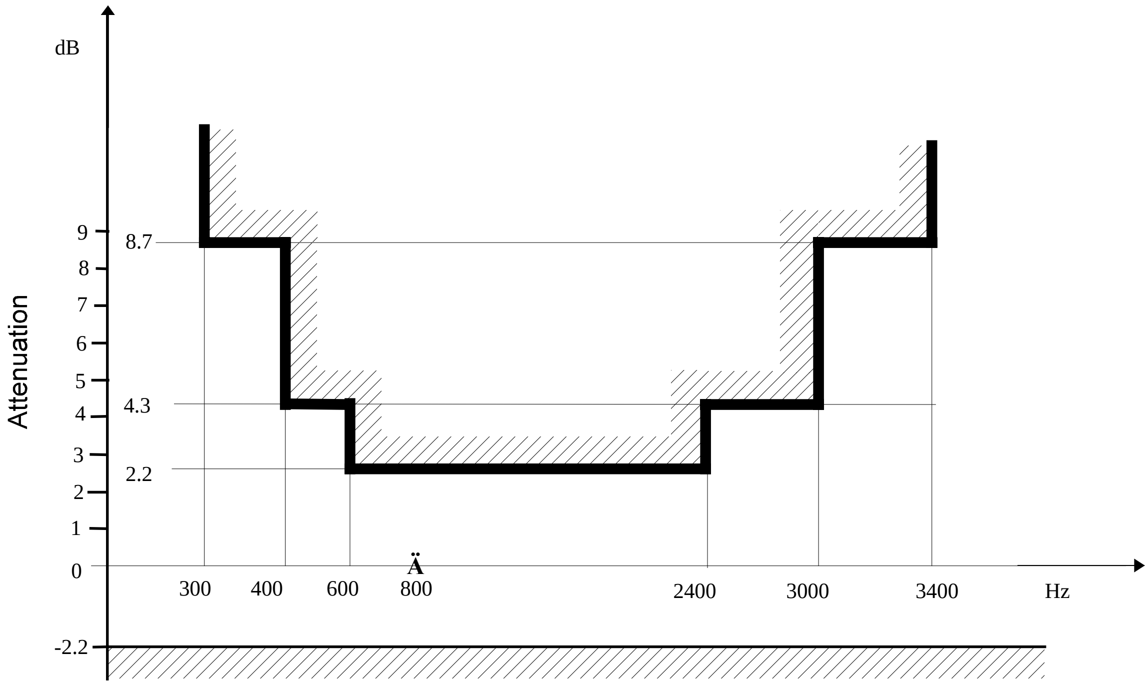


Figure 1/G.132

Permissible attenuation variation with respect to its value measured at 800 Hz
(objective for worldwide 4-wire chain of 12 circuits in terminal service)

f) Group delay distortion

The group delay variations for a similar world-wide connection should not exceed the values shown below (CCITT Recommendation G.133)

Table 1/G.133

	Lower limit of frequency band (ms)	Upper limit of frequency band (ms)
International chain	30	15
Each of the national 4-wire extensions	15	7.5
On the whole 4-wire chain	60	30

Note - Limits given in Table 1/G133 should be met both for analogue circuits and mixed circuits with analogue and digital sections.

g) Absolute delay

The planning values for the mean one-way propagation time are given in CCITT Recommendation G.114 and are reproduced in the following table:

Transmission medium	**Contribution to one-way propagation time**	**Remarks**
Terrestrial coaxial cable or radio relay system; FDM and digital transmission	4µs/km	Allows for delay in repeaters and regenerators
Optical fibre cable system; digital transmission	5µs/km	Allows for delay in repeaters and generators
Submarine coaxial cable system	6µs/km	
Satellite system - 14 000 km altitude - 36 000 km altitude	 110 ms 260 ms	Between earth stations only
FDM channel modulator or demodulator	0.75 ms [a]	
FDM compandored channel modulator or demodulator	0.5 ms [b]	Half the sum of propagation times in both directions of transmission
PCM coder or decoder	0.3 ms [a]	
PCM/ADPCM/PCM transcoding	0.5 ms	
Transmultiplexer	1.5 ms [c]	
Digital transit exchange, digital-digital	0.45 ms [d]	
Digital local exchange, analogue-analogue	1.5 ms [d]	
Digital local exchange, analogue subscriber line-digital junction	0.975 ms [d]	
Digital local exchange digital subscriber line-digital junction	0.825 ms [d]	
Echo cancellers	1 ms [e]	

a) These values allow for group-delay around frequencies of peak speech energy and for delay of intermediate higher order multiplex and through-connecting equipment.

b) This value refers to FDM equipments designed to be used with a compandor and special filters.

c) For satellite digital communications where the transmultiplexer is located at the earth station, this value may be increased to 3.3 ms.

d) These are mean values: depending on traffic loading, higher values can be encountered, e.g. 0.75 ms (1.950 ms, 1.350 ms or 1.250 ms) with 0.95 probability of not exceeding. (For details, see Recommendation Q.551.)

e) Echo cancellers, when placed in service, will add a one-way propagation time of up to 1 ms in the send path of each echo canceller. This delay excludes the delay through any codec in the echo canceller. No significant delay should be incurred in the receive path of the echo canceller.

h) Stability and echo (see also CCITT Recommendations G.122 and G.131)

Referring to Figure 8.8, to avoid instability during all normal conditions of call set-up, clear-down, call transfer, etc., of a connection, the distribution of the loss (a-b) during the worst situation should be such that the risk of this loss reaching 0 dB or less should not exceed 6 in 1,000 calls. This limit can be met by imposing the following simultaneous conditions on the national network:

- the sum of the nominal transmission losses in both directions of transmission a-t and t-b should not be less than (4 + n) dB where n is the number of 4-wire circuits in the national chain;

- the stability balance return loss at the terminating set should have a value of not less than 2 dB for normal operation conditions;

- the standard deviation of variations of transmission loss of a circuit should not exceed 1 dB.

The loss (a-b) is then (2 + 4 + n) = (6 + n) dB and the standard deviation $2n^{0.5}$ dB.

In addition, to provide sufficient loss (a-b) when the call is established, the mean value of the stability balance return loss should be at least 6 dB and the standard deviation not more than 2.5 dB.

The echo loss (a-b) on an established connection should have a distribution of mean value not less than $(15 + n)$ dB with a standard deviation of $(9 + 4n)^{0.5}$.

Improved balance return losses at t are obtained with digital local exchanges, when the local line is permanently associated with the 2w/4w conversion unit.

i) Linear crosstalk

The limits given in CCITT Recommendation G.151 should be respected, i.e.:

- between circuits: at least 65 dB;

- between go and return channels: at least 43 dB.

j) Quantising distortion

The incorporation of unintegrated digital processes in international connections can result in an appreciable accumulation of transmission impairments. The limits for this type of impairment are expressed in 'quantising distortion units' or qdu and are specified in CCITT Recommendation G.113.

The planning objective for an international telephone connection is that the quantity 5 + 4 + 5 = 14 qdu should not be exceeded, i.e. each national segment is allowed a contribution of 5 qdu and the international portion up to 4 qdu.

The planning values for quantising distortion are given in the following table:

Planning values for quantising distortion
(Speech service only; see § 4 for voiceband data considerations)
(see Notes 1, 11 and 12)

Digital process	Quantising distortion units (qdu)	Notes
Process involving A/D conversion		
8-bit PCM codec-pair (according to Recommendation G.711[2], A- or µ-law)	1	2, 3
7-bit PCM codec-pair (A- or µ-law)	3	3, 4, 5
Transmultiplexer pair based on 8-bit PCM, A- or µ-law (according to Recommendation G.792)	1	3
32 kbit/s ADPCM (with adaptative predictor) (combination of an 8-bit PCM codec pair and a PCM-ADMPCM-PCM tandem conversion)	3.5	6
Purely digital processes		
Digital loss pad (8-bit PCM, A- or µ-law)	0.7	7
A/µ-law or µ/A-law converter (according to Recommendation G.711 [2])	0.5	10
A/µ/A-law tandem conversion	0.5	
µ/A/µ-law tandem conversion	0.25	
PCM to ADPCM to PCM tandem conversion (according to Recommendation G.721)	2.5	8, 9
8-7-8 bit transcoding (A- or µ-law)	3	9

Note 1 - As a general remark, the number of units of quantising distortion entered for the different digital processes is that value which has been derived at a mean Gaussian signal level of about - 20 dBm0. The cases dealt with in Supplement 21 (at the end of this fascicle) are in accordance with this approach.

Note 2 - By definition.

Note 3- For general planning purposes, half the value indicated may be assigned to either of the send or receive parts.

Note 4 - This system is not recommended by CCITT but is in use by some Administrations in their national networks.

Note 5 - The impairment indicated for this process is based on subjective tests and was provided by Study Group XII.

Note 6 - For this 32 kbit/s ADPCM process a value of 3.5 units was derived by Study Group XII from subjective measurements on a combination of an 8-bit PCM codec par and a PCM/ADPCM/PCM conversion according to Recommendation G.721.

Note 7 - The impairment indicated is about the same for all digital pad values in the range 1-8 dB. One exception is the 6dB A-law pad which introduces negligible impairment for signals down to about - 30 dBm0 and thus attracts 0 units for quantising distortion.

Note 8 - The value of 2.5 units was derived by subtracting the value for an 8-bit PCM codec pair from the 3.5 units determined subjectively for the combination of an 8-bit PCM code pair and a PCM/ADPCM/PCM conversion. Multiple synchronous digital conversions, such as PCM/ADPCM, PCM/ADPCM/PCM, are assigned a value of 2.5 units.

Note 9 - This process might be used in a digital speech interpolation system.

Note 10 - The qdu contribution made by coding law converts (e.g., µ-law to A-law) are assigned to the international part.

Note 11 - The qdu assignments to these digital processes reflect, to the extent possible, only the effect of quantisation distortion on speech performance. Other impairments, such as circuit noise, echo and attenuation distortion on speech performance. The effect of these other impairments must therefore be taken into account in the planning process.

Note 12 - The qdu impairments in this table are derived under the assumption of negligible bit error.

k) Jitter

The limits given in Table 3 of CCITT Recommendation G.921 (reproduced below) should be respected. The limits for jitter at a hierarchical interface are given in CCITT Recommendation G.823.

Table 3/G.921

The maximum output jitter in the absence of input jitter for a digital section

(Measurements are made in accordance with the method shown in Figure 1/G.823)

Bit rate (kbit/s)	HRDS length (km)	Maximum output jitter for a digital section		Measurement filter bandwidth		
		Low frequency limit ($f_1 - f_4$) unit interval peak-to-peak	High frequency limit ($f_3 - f_4$) unit interval peak-to-peak	Band-pass filter having a lower cut-off frequency f_1 or f_3 and an upper cut-off frequency f_4		
				f_1	f_3	f_4
2 048	50	0.75	0.2	20 Hz	18 kHz (700 Hz)	100 kHz
8 448	50	0.75	0.2	20 Hz	3 kHz (80 kHz)	400 kHz
34 368	50	0.75	0.15	100 Hz	10 kHz	800 kHz
34 368	280	0.75	0.15	100 Hz	10 kHz	800 kHz
139 264	280	0.75	0.075	200 Hz	10 kHz	3 500 kHz

Note - For interfaces within national networks the frequency values (f_1 and f_3) shown in parenthesis may be used.

8.4.15 Local networks

Local network planning should always envisage networks using shorter subscriber lines with the smallest possible gauge of conductor. These gauges should preferably be limited to two types, in the interests of standardisation, and obviously have to be selected in particular cases to meet the LR requirements specified above. The LRs of the telephone sets therefore have to be known with some precision. These sets should be provided with automatic transmission regulation (ATR), and the SLR/RLR must be clearly specified by the supplier. Telephone sets equipped with ATR in general use more sensitive transmitter and receiver elements, the regulator being controlled by the loop current to reduce the sensitivity on short subscriber loops. Special precautions should therefore be taken when regulated sets are used with a PABX as the high feeding current could result in a sensitivity which does not correspond to the line length to the local exchange. PABX suppliers should be required to specify the measures to be taken to adjust the feeding current in these cases.

To meet the requirements of digital switching equipment the maximum resistance of the subscriber loops should be the value specified by the manufacturer. The resistance and capacitance per unit length of cable used must be known so that, for planning purposes, the LR can be calculated for the line plus telephone set.

In general, local area networks should be extended by the use of RSUs, DSM, etc., to reduce the lengths of the subscriber lines - in particular in rural areas. Single channel subscriber radio terminals and TDMA terminals should, whenever practicable, be installed at local exchange or RSU sites. If not, they must be connected to these sites on a 4-wire basis (audio or PCM respectively). The LRs of the local lines for these systems must conform to the transmission plan. This applies also to PABX connections.

The local networks have to be suitable for the use of appropriate data terminals so that the terms of, among others, CCITT Recommendations V.2 (levels) and V.23 (loss) have to be met.

It is anticipated that metallic local lines will continue to be used for the basic rate access to any ISDN which may be established. The requirements for the pairs in the main and subscriber distribution cables which have to provide this access are given in CCITT Recommendation G.961.

Details of the digital sections for basic and primary rate access to the ISDN are given in ITU-T (formerly CCITT) Recommendations G.960 and G.962 respectively.

8.5 Network synchronisation plan

8.5.1 Introduction

For many reasons, not the least significant of which is the virtual unavailability of analogue systems, there is an increasing use of digital switching and transmission systems in practically all telecommunication networks. These digital facilities operate in parallel, or in many cases as digital islands, with the original analogue systems until these are withdrawn from service and replaced.

While comprehensive synchronisation facilities are not strictly necessary for a purely telephony service over digital connections - for example digital islands can operate in a plesiochronous mode using oscillators (clocks) of only moderate stability - the future interconnection of these islands and the use of the network for non-voice services imposes a requirement to prepare for network synchronisation at an early stage. As explained later, plesiochronous operation with a high stability is extremely expensive and it is preferable to operate in a synchronous mode using inexpensive slave clocks and a highly stable reference source.

For Administrations embarking on a digitalisation programme, it is important to consider the synchronisation requirements at an early stage. In particular, it is essential to ensure that all digital switching equipment is provided with facilities for any type of synchronisation to be implemented when required.

The planning objectives for slip rates and the commonly used synchronisation methods are considered in the following sections.

8.5.2 Purpose of synchronisation

At each digital exchange of the network, the digital signals are received at their nominal incoming rate (bit rate) and stored in a buffer, i.e. a time slot buffer or frame buffer accommodating 8 and 256 bits in the case of the CEPT 2 048 kbit/s hierarchy or 8 and 193 bits in the case of the North American 1 544 kbit/s hierarchy inputs. The stored bits are then switched at a frequency determined by the exchange master oscillator, or "clock". If, for example, the incoming bit rate is higher than that of the exchange clock, there is obviously no time to transmit the stored bits before another bit stream arrives. Similarly, if the incoming bit rate is lower than that of the clock, the stored bits will be sent twice before the arrival of a new bit stream. In other words, the incoming bit rate, which is determined by the clock at the sending exchange, must be equal to that of the clock at the receiving exchange if transmission disturbances are to be avoided. These disturbances are known as "slips" and the synchronisation of the exchange clock frequencies throughout the digital network aims to reduce or eliminate slips due to frequency differences. The slip rate, i.e. the number of slips over a specified time is in fact directly proportional to this difference in frequency. It should be noted, however, that slips can occur due to other causes, for example phase jitter on transmission links, which can be reduced by suitable buffer circuit design.

8.5.3 Planning objectives

In general, the planning objective for network synchronisation is stated as a maximum permissible slip rate, that is to say the maximum number of slips allowed to occur over a specified time interval - for example one hour, one day, etc. This objective can be determined either from a knowledge of the acceptable slip rate for the most sensitive service likely to be provided on the network or by the use of a general standard recommended by the CCITT as being adequate for most of the offered services.

a) Objective derived from most sensitive service

The effects of slips on the different telecommunication services provided by means of a digital network depend on the coding redundancy of the digital signal and on its bit rate. The higher the coding redundancy, the less the effect of a slip; the higher the bit rate, the greater the impact of a slip. For example:

- PCM coded speech is rather insensitive to slips due to its high coding redundancy;
- PCM coded voiceband data modems could lose synchronisation due to the phase shifting due to slips;
- data transmission over a 64 kbit/s circuit has significantly less coding redundancy than PCM speech and is therefore more sensitive to slips;

- facsimile, depending on the coding technique, can be affected by a single slip which, in the worst case, could destroy the whole picture;
- telephony using common channel signalling can be designed so that the probability of a signal being lost or misinterpreted due to slip is low, in general resulting only in a slight delay of the telephony signal which does not normally affect the signalling function.

Of the services considered above, the most sensitive to slip have been found to be the intermediate rate data services which can only tolerate a slip rate of about 12 slips in 24 hours. Thus, to ensure a satisfactory slip performance, a planning objective much better than the tolerable level has to be defined, an appropriate objective being not more than 5 slips in 24 hours over the digital hypothetical reference connection (HRX) which includes up to 13 switching nodes.

b) CCITT recommended slip rate objectives

The basic data for the definition of a general slip rate objective can be found in CCITT Recommendations G.811 and G.822, the first Recommendation being concerned with the specification of the reference clocks at international network nodes (ISC) and the second with the practical objectives for the various segments of a complete international 64 kbit/s ISDN connection, the HRX.

CCITT Recommendation G.822 includes objectives from which the slip rates per inter-exchange link of the HRX can be derived, the information contained in this Recommendation being summarised below:

i) The end-to-end slip rate performance should satisfy the requirements for telephone and non-telephone services on a 64 kbit/s digital connection in an ISDN.

ii) The slip rate objectives for the above are stated in terms of the digital hypothetical reference connection (HRX), specified in CCITT Recommendation G.801, of 27 500 km length.

iii) It is assumed that the international switching centres (ISC) are interconnected by means of links which operate in the plesiochronous mode using clocks with the accuracies specified in CCITT Recommendation G.811. (In this Recommendation the slip rate objective per link is 1 slip in 70 days.)

iv) For the definition of a practical slip rate objective it is recognised that the slip rate may considerably exceed the value computed for a series of plesiochronous links due to various design, environmental and operational conditions in the international and national segments concerned, for example temporary loss of synchronisation.

v) Threshold values of slip rate are defined as a compromise between desired performance and normally achievable performance according to the following table:

Acceptable controlled slip performance on a 64 kbit/s international connection or bearer channel

Performance category	Mean slip rate	Proportion of time (Note 1)
(a) (Note 2)	≤ 5 slips in 24 hours	> 98.9%
(b)	> 5 slips in 24 hours and ≤ 30 slips in 1 hour	< 1.0%
(c)	> 30 slips in 1 hour	< 0.1%

Note 1 - Total time ≥ 1 year.
Note 2 - The nominal slip performance due to plesiochronous operation alone is not expected to exceed 1 slip in 5.8 days. See § 2.4.

In the above table, performance category a) represents a satisfactory slip rate. Slip rates in category b) are considered to be degraded while those in category c) are unacceptable.

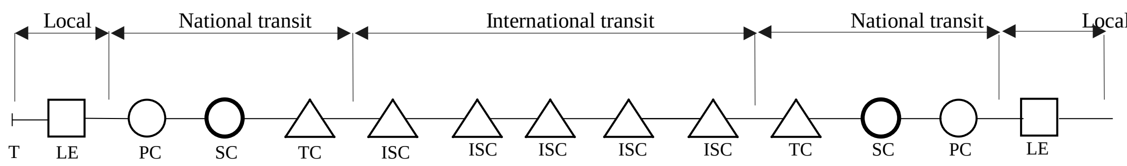
vi) The allocation of slip rate objectives to the various segments of the HRX is based on the subdivision of the percentage of time objectives for categories b) and c) above as shown in the following table:

Allocation of controlled slip performance objectives

Portion of HRX derived from Figure 1/G.801 [1]	Allocated proportion of each objective in Table 1/G.822	Objectives as proportion of total time	
		(b)	(c)
International transit portion	8.0%	0.08%	0.008%
Each national transit portion (Note 2)	6.0%	0.06%	0.006%
Each local portion (Note 2)	40.0%	0.4%	0.04%

As the impact of slips over different segments will vary in importance according to the type of service and the level of traffic affected, the allocation places stricter limits on the international and national transit exchanges than on local exchanges. In the above allocations advantage is taken of the fact that the probability of two or more segments of the HRX experiencing simultaneous excessive slips is low.

vii) The subdivision of the HRX for the purpose of allocating slip rate objectives is shown below:



(FIGURE 1/G.822)

Subdivision of the HRX for the purpose of allocation of slip performance objectives

8.5.4 Interpretation of CCITT Recommendations

An international digital connection can consist of up to 13 nodes, as shown on the HRX diagram. If all 13 nodes were operating according to CCITT Recommendation G.811 (plesiochronous mode), each of the 12 links would have the specified, theoretical slip rate of 1 per 70 days so that the overall connection would experience 1 slip per 70/12 days, or 1 slip per 5.8 days.

As a fully synchronised network theoretically experiences no slips, the interconnection of two such synchronised national networks by means of the four international links of the HRX would therefore result in an overall theoretical slip rate of 1 slip per 70/4, or 1 slip per 17.5 days.

However, for the reasons given in the previous section and in particular during temporary loss of synchronisation, a slip rate significantly in excess of the above theoretical values will be experienced over a practical connection. The limits defined in the CCITT Recommendations are intended to take these practical conditions into account. Thus 46% of the mean annual slip rate of the HRX is allocated to each national segment, which means that each of these segments is allowed to contribute 2.3 slips per 24 hours to the mean HRX slip rate of 5 per 24 hours. This mean value has to be maintained for 98.9% of the time over the HRX.

In terms of the national exchange links the objectives for slip rate can be expressed as follows:

National transit exchange:	1 slip per 7 days
Local exchange:	1 slip per 12 hours

The objective for each international exchange link is already defined in CCITT Recommendation G.811 (1 slip per 70 days).

The above objectives are used to derive the frequency accuracies required for the exchange clocks at the relevant network levels, the accuracy of the ISC clock being defined (10^{-11}) in CCITT Recommendation G.811.

8.5.5 Frequency control (synchronisation) methods

Two basic methods can be used to control the frequencies of the exchange clocks within their prescribed limits:

- the plesiochronous (i.e. almost synchronous) method, in which the exchange clocks are completely independent but running at the same frequency with very high long-term stability;
- the synchronous method, in which all the exchange clocks within the digital network (or sub-network) are interconnected and controlled to run at the same frequency, ideally at that of a stable reference source.

Plesiochronous operation, in practice, is normally only used on the international digital network, each ISC being equipped with a caesium clock - which at present is the only reference source available which can provide the frequency accuracy (10^{-11}) and inherent long-term stability required to meet the slip rate objective defined in CCITT Recommendation G.811.

Synchronous operation avoids the use of expensive caesium clocks at each exchange as, by definition, this method is intended to reduce or eliminate the frequency difference between the exchange clocks. The clocks used on a synchronous network are relatively simple crystal oscillators which, while their long-term stability is poor, have the advantage of being easily and continuously controlled in frequency by the application of a variable voltage. This type of clock is therefore known as a voltage-controlled crystal oscillator, or VCXO. In view of the poor long-term stability, the absolute frequency of a synchronised network using VCXOs will drift with time although the measured slip rate will be negligible. To maintain the absolute frequency, it is therefore necessary to use a reference source of high long-term stability, in practice the caesium clock of the ISC when digital international operation is required.

The two most appropriate methods used for synchronous operation of a digital network are:

- the master-slave method, is which the clock at only one exchange (for example the ISC or a high-level transit exchange) functions as the reference source with all other exchange clocks phase-locked ("slaved") to this reference;
- the mutual single-ended control method, in which every exchange clock is locked to the average of all incoming frequencies so that the network is compelled to operate at the same frequency.

Both these methods rely on some form of regulator to control the frequencies of the VCXOs, often the so-called "proportional plus integrating" (PI) type.

In a mutually synchronised network two conditions must be observed; all the regulators must be of the same type and one of the exchanges must be unregulated to act as a sink to accommodate the total phase error of the network and prevent frequency drift. This unregulated source is the frequency reference of the network and, for obvious reasons, should normally be the clock of the ISC which is equipped with buffers to take care of short-term phase variations.

The master-slave method is normally used on relatively small star networks and can continue to be used at the lower network levels as the configuration develops. When this development reaches the stage at which mesh-connected national transit exchanges are introduced, mutual single-ended control can be implemented, with the reference source at the unregulated ISC or other designated high-level node. Thus in a fully developed network all three methods will be used; plesiochronous operation at the ISC, which also serves as the reference source for the national network, mutual synchronisation of the mesh-connected national transit exchanges and master-slave synchronisation at the local and lower network levels. These methods, as applied to a typical network, are shown schematically in Figure 8.10

Due to its relative simplicity, the master-slave method of synchronisation is often used on large networks in the hierarchical master-slave mode. In this system, the caesium clock equipped at the international exchange is duplicated at a selected national transit exchange (or of course at a second international gateway if applicable). In the case of failure of the master clock, the clock at the secondary centre takes over this function - thus improving the reliability.

A further refinement to the master-slave method is the so-called pre-selected alternative method in which each slave exchange is assigned a number of incoming routes from which its reference signal is selected according to a pre-set programme, changeover to the next alternative incoming reference would occur on loss of signal or on some other clear indication of a fault on the original incoming signal.

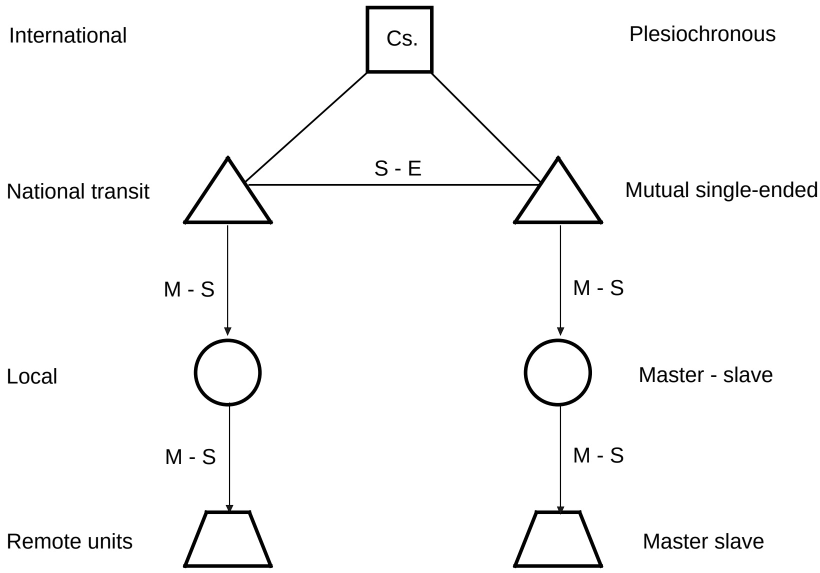


Figure 8.10: Synchronisation methods

8.5.6 Interworking between analogue and digital exchanges

The digital links used for the interconnection of analogue and digital exchanges are equipped with PCM terminals at the analogue ends. The clocks at the sending and receiving sides of these terminals are normally independent but when connected to a digital exchange, the frequency of the multiplex receive clock is determined by that of the digital exchange so that the send side clock has to be synchronised. This is achieved by looping at the PCM terminal as shown in Figure 8.11, resulting in both PCM clocks being effectively slaved to the digital exchange.

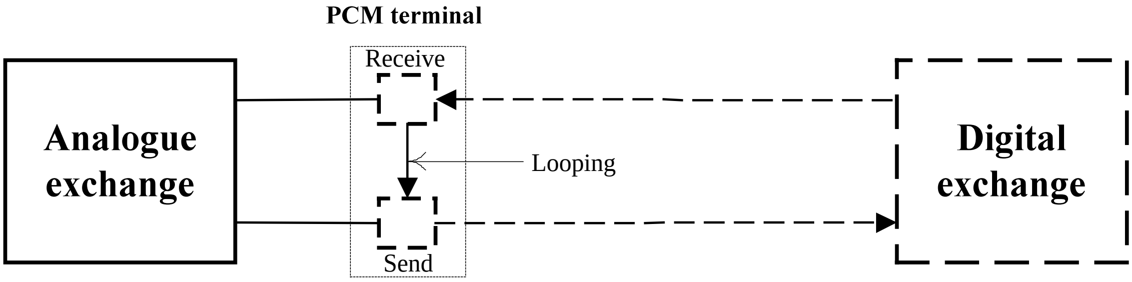


Figure 8.11: Cooperation between digital and analogue exchanges

8.5.7 Summary of objectives, clock accuracies, etc.

a) Planning objectives for slip rate

In general, for a fully synchronised national network, the contribution to the mean slip rate of any international digital connection should not exceed that defined in CCITT Recommendation G.822, that is to say not more than 46% of the total mean slip rate of 5 slips per 24 hours. This corresponds to less than 2.3 slips per 24 hours and should not be exceeded for more than 1% of the time.

To meet this objective, the slip rates for the inter-exchange links of the international connection should not exceed the following values:

International link:	1 slip/70 days (CCITT G.811)
National transit link:	1 slip/7 days
Local exchange link:	1 slip/12 hours

b) Exchange clock accuracies

The long-term frequency accuracies (stability) required to meet the above slip rate objectives over the fully synchronised national network are as follows:

Exchange clock	Accuracy
International exchange	10^{-11}
National transit exchange	10^{-10}
Local exchange	10^{-9}
RSU/concentrator	10^{-7}

These accuracies are required for short-term synchronisation failures when a particular clock will be free-running until synchronisation is restored.

If the national reference is not a caesium clock, the frequency of the reference source should be checked every six months.

In the fully synchronised configuration the reference clock should be located in the ISC.

The frequency accuracy of all PCM terminal clocks used on the network should be in accordance with the relevant CCITT Recommendations (G.700-series).

8.5.8 Distribution of timing signal

The reference timing signal is distributed over the national network by the inter-exchange links in use. In the early stages of digitalisation these could consist of analogue links using the reference pilot system.

In any case, for reliability, two or more timing links should be provided to each digital node.

8.5.9 CCITT/ITU-T Recommendations

Reference should be made to Recommendations G.810, G.811, G.812 and G.822.

8.5.10 Satellite networks

The synchronisation of systems incorporating satellites is considered on the ITU Handbook on satellite communications (Geneva 1988) to which reference should be made.

8.6 Charging plan

8.6.1 General considerations

The financial charges to users of telecommunications networks can be distinguished in a general way in two categories:

a) fixed charges;

b) usage-dependent charges.

In a development plan, which is mainly concerned with the Public Switched Telephone Network (PSTN) and its future development into a type of Integrated Services Digital Network (ISDN), for the category of the fixed charges can be mentioned the connection fee (which may include a guarantee), rentals for the network access and for terminal equipment and possible flat-rate arrangements for certain categories of communications (e.g. local calls) or for certain quantities of communication (for example a number of free charging units per period, but included in the rental).

As far as usage-dependent charges are concerned, this refers to the cost per call or communication, and this type of charging has to be performed by the telecommunication equipment. In general this will include local calls, various types of long-distance (trunk) calls, international calls, special service calls and calls to special networks and facilities.

The tariffs for the fixed charges as well as for the usage-dependent charges have to be fixed in such a way that they cover the cost of the corresponding network part, and in a general way that the overall financial result meets the objectives of the operating entities. However, in view of socio-economic considerations, it may be desirable to deal in a different way with certain categories of users, for example those connected to costly rural telecommunication networks, in providing reasonable rates for these services, which may financially be supported from additional revenue from other services, such as long-distance and international traffic.

This chapter will concentrate on the functional requirements of the usage-depending charges on the telecommunications equipment functions involved, i.e. switching equipment and signalling systems, and the possible arrangements for these functions, both for automatic and operator assisted calls.

The issue of the different rate levels will be dealt with in the section concerning financial aspects, which will cover the actual rate structure and rate levels of the existing services as well as the proposed rate structure and rate levels for existing and new services, as a result of the total financial analysis of the telecommunication services.

8.6.2 Basic characteristics of usage-depending charging

In general only successful communications should be charged, i.e. when a response (answer signal) is received from the called subscribers.

Normally the calling subscriber will be charged for the calls/communications; however in some exceptional cases like free-phone service, calls may be charged to the called subscriber.

The type of charging applied to calls/communications can be classified as follows:

a) free of charge;

b) fixed charge per call;

c) time-dependent charge;

d) a combination of b) and c).

As far as calls free of charge are concerned, this applies to certain special services belonging to Administration/Telecom Operator, and to free-phone service, at least for the calling subscriber.

A fixed charge per call is (and was) in many networks applied to local calls and to calls to certain special services belonging to the Administration/Telecom Operator.

Time-dependent charging will in any case be applied to long-distance (trunk) and international calls, to calls for special networks and facilities, but may also be applied to local calls and certain special service calls belonging to the Administration/Telecom Operator.

The chargeable duration is defined as the period between the response of the called subscriber (answer signal) and the end of the communication (clearing signal).

The chargeable duration can be measured on the basis of real charged time or on the basis of charging units.

The first method can be applied with detailed charge recording systems, providing at least time, call duration, charge and called subscriber number for each call.

The second method can be provided by periodic pulse metering (PPM), which gives in general one charge unit at the answer by the called subscriber, followed periodically by other charge units. The duration of the period between charge units defines the rate level of the time dependent charging.

This charging method is used for bulk charging which adds the charge units of all communications together on one register (meter) for each subscriber.

However, this method is also used for detailed charge recording, where instead of the call duration the number of charge units per call is presented.

8.6.3 Equipment considerations

The requirements for the provision of the usage-dependent charges for calls/communications have to be translated into requirements for switching equipment, signalling systems and charge recording and billing facilities, and in a general way these requirements can be formulated as follows:

a) charge rate definition by analysis of the origin (if needed) and destination of calls;

b) provision of the information on the charges per call, i.e. chargeable duration or number of charging units;

c) provisions for charge registration by means of registers (meters) per subscriber for bulk registration of charging units and/or provisions for detailed charging on centralised storage with information about time, duration or number of charging units per call and the called subscriber number;

d) presentation of charging results for subscriber billing.

Charging functions in a network can be centralised or decentralised, depending on technical and economical considerations. In networks with old electromechanical exchanges using periodic pulse metering, generally local charging is performed in the local exchange, national long-distance charging in the originating trunk exchange and international charging in the international exchange. With the more intelligent SPC-exchanges there is a tendency to centralise these charging functions, for example for all national traffic in the local exchange.

Regarding the requirements mentioned under a), the analysis of the number of digits of the destination, required for charging purposes, should be limited as much as possible, and the numbering system should be developed in such a way that it will be easy to distinguish destination charging zones.

As far as origin analysis is concerned, this may be required for a charging system with long-distance charging in the originating trunk exchange, and where the primary area served by this trunk exchange will be composed of a number of charging areas. In such a case the trunk exchange has to detect from which charging area the call is originated.

The origin of the call has also to be transmitted in the case of a centralised detailed charging system, for example for a centralised detailed charging system on the level of trunk exchanges for national long-distance traffic, the calling subscriber number has to be transmitted from the local exchange to the trunk exchange by means of appropriate signalling arrangements.

In the case of decentralised pulse-metering systems, i.e. charging of long-distance calls in trunk exchanges and for international calls in the international exchange, the metering signals have to be transmitted over circuits to the local exchange, and the appropriate arrangements have to be made in the line (supervisory) signalling to provide for metering signals during conversation (outband signals). The transmission of metering signals may be extended over subscriber lines to terminal equipment such as a PABX, payphone or meter at subscriber premises.

Recording of charge information will be provided on stores or mechanical meters, on-line for bulk charging of metering units, and on magnetic tape units or hard disks for detailed charging systems. In case of common storage equipment, redundancy arrangements have to be provided to ensure reliable storage of this important information.

Finally, for the preparation of billing information of the subscribers, appropriate arrangements have to be made, which consist of photographing or manual or automatic reading of mechanical meters and subsequent processing of the charging information on mass-storage devices (magnetic tape or disk), and the subsequent processing of information on these mass-storage devices for billing information for the subscribers. Very often this last activity is undertaken by an outside specialised firm.

8.6.4 Review and possible revision of existing charging plans

A review and possible revision of existing charging plans should include an analysis of the basic charging methods applied to the different rates as well as the entire rate structure.

However, whatever modification will be proposed, it will require a solid financial analysis with detailed knowledge of traffic volumes and traffic distribution in order to predict in a reliable way what the effect will be on the revenues expected with revised rates.

As far as basic charging methods are concerned, there are still many cases where local calls are charged with one charging unit, independent of the holding time of the calls, and time dependent charging may be considered for this type of service.

However, with periodic pulse metering systems, which are used for the different categories of traffic in a network and also for international traffic, there is an interdependence of rates, and this is the charging unit which can be transmitted over circuits at approx. 2 units per second at the most. With an inexpensive charging unit, it may therefore be difficult or even impossible to realise expensive international rates, and, therefore, it may be useful to initially raise the local rates by increasing the price of the basic charging unit and introduce time-dependent charging of local calls later.

A question related to periodic-pulse metering systems is the accuracy of the first interval between the charging pulse on answer and the first periodic charging pulse sent to the charge recording equipment.

If the system applied considers every pulse sent as a charging pulse, then the average duration between the charging pulse at answer and the first periodic charging pulse will be 1½ period (assuming that the first periodic charging pulse is suppressed).

Another possibility is to use time pulses and count a number of pulses before a charging pulse is sent. In this way it is possible to reduce the time between the charge pulse at answer and the first periodic charging pulse to a value close to the real period, which, of course, is an advantage for the Telecom operator from the revenue point of view.

Another question concerning the basic rates is the possibility to have a differentiation in time-dependent rates according to the time of the day (day-night/weekend tariffs).

The possible arrangements with such a system may vary from a simple two-stage rate system with normal (high) rates during the day and reduced rates during night, weekend and holidays, to a system with more differentiation, for example including also some intermediate rates for other specified periods of the day and thus providing three or four rate levels per tariff category, each of them applicable during certain periods of the day.

For the purpose of definition of long-distance rates, the national network is divided in charging areas. The charging areas will cover a group of local networks and coincide very often with a switching configuration, for example the originating trunk exchange (group centre or primary centre) with its dependent local exchanges.

For trunk calls within the charging area, one rate applies, independent of distance.

For trunk calls to other charging areas, two different methods can be applied for definition of the charging rates:

- in smaller networks, by making a distinction between adjacent and non-adjacent charging areas;

- by making a distinction based on the distance between charging areas, and in this case a geographical charging point has to be defined for each charging area; normally, this will be the trunk exchange, provided that its position is more or less in the middle of the charging area, otherwise a different point has to be selected.

A problem with these rates is that calls within a charging area may cover a considerable distance and will be charged at a lower rate than calls to an adjacent area and covering a small distance, for subscribers living close to a charging area border. For subscribers, it is difficult to understand why calls over a short distance should be charged at a higher rate than calls over a longer distance.

This problem can be solved by applying the same rate for intra-area calls as for calls to adjacent charging areas, provided that the geographical size of the charging areas in a network is not too large.

Another issue is the number of long-distance rates applied in a network. Very frequently, the long-distance charging plan has been designed by converting the manual rate system into a system for the automatic network, and this may have resulted in a system with many different long-distance rates. Since the cost of transmission facilities is not strictly proportional to distance it could simplify the charging if the rates beyond for example 500 km were identical.

The general tendency is to reduce the number of different rates in a national network, and during the review of the actual system such measures might be proposed.

Once again, it has to be emphasised that such measures can only be introduced after a detailed revenue study based on detailed information about traffic volumes and traffic distribution.

For international traffic, similar arguments can be mentioned as for national long-distance traffic, there is a tendency to simplify the rate structure for international traffic since the cost factor related to distance is becoming less important. This is to a large extent due to the development of satellite communications and the introduction of large capacity optical fibre (submarine) cable systems, which have changed considerations of cost and distance considerably.

Requirements for international accounting also have to be reviewed since their requirements are closely related to those for charging of international traffic.

Consideration has to be given to charging requirements for traffic to special networks and facilities which may require special charging methods and rate levels not included in the existing charging facilities in the network.

In view of the previously mentioned considerations, it will be useful to review the charging requirements for the different types of traffic and to define at which network levels these functions should be implemented. These considerations also have to include issues such as charge registration and data processing for billing purposes.

The question of revenue sharing is now becoming increasingly important due to the presence of more than one telecommunication operating organisation in many (including developing) countries.

Thus it will be necessary for the various service providers to exchange call data, to share billing information and to coordinate customer billing. This problem is already faced in countries in which independent cellular radio networks exist.

In some rural areas, privately franchised PCOs are coming into operation, the billing function being automated by the national Administration. Call charges are processed at the time the call is made, collection is made by the PCO franchisee and call data are automatically transferred to the national network operator for future settlement.

An extension of this principle can be found in the so-called 'phone shops', established in South Africa for example, which consist of up to 10 public pay-phones above each of which is mounted a display and monitoring unit showing the progressive cost of the call. The licensee of the phone shop is provided with a control unit to which each display is linked in a bus configuration, a printer also being provided for the control unit. The caller is given the choice of either pre-payment or payment on call completion. printed receipts being supplied to callers. In this application, the phone shop owner reconciles his accounts payable to the service provider by submitting printouts of transactions at regular intervals. The margins and rates for call set-up are pre-set within agreed limits.

8.6.5 References

1. CCITT Blue Book, Volume II, Fascicle II.1, General Tariff principles, charging and accounting in international telecommunications services, Geneva, 1989.

2. GAS 5: Economic studies at the national level in the field of telecommunications, 1976.

3. GAS 6: Handbook on economic and technical aspects of the choice of telephone switching systems, 1981.

4. CCITT Manual "National telephone networks for the automatic service". Geneva, 1964.

8.7 Network security plan

8.7.1 Introduction

Network digitalisation almost inevitably results in a reduced number of higher capacity switching nodes connected by fewer, and high capacity, transmission systems. Major failures therefore affect an increased number of circuits and it is now even more important to take adequate precautions to minimise the effects of these failures, as well as, if possible, to reduce the fault rate itself.

If these precautions are not taken, the resulting congestion can cause, in addition to inconvenience to the subscribers, a serious loss of revenue to the network operator.

The guidelines given in this section should enable the planning personnel to prepare:

- an assessment of the current network with particular emphasis on its vulnerability;
- proposals for improvements to the resilience of the network to major interruptions;
- proposals to include security precautions in all future network extensions and modifications;
- an analysis of the cost of providing network security for several scenarios,

for which the following inputs are required:

- routing plan of current network;
- routing plans from optimisation of future network;
- fault reports.

This section provides a brief review of the measures which can be taken to improve network security, including some elementary measure to improve reliability in general.

The methods which could be available for future networks are also briefly reviewed.

8.7.2 General

As a result of the increasingly large-scale use of digital switching and transmission, telecommunication networks are now tending to consist of a reduced number of higher capacity exchanges connected by high-capacity transmission systems - often optical fibre cable systems of extremely high capacity. It is well known that digital switching systems are subject to virtually complete interruption on the failure of the central processing unit, in contrast to older systems in which the failure of a single element rarely resulted in a complete failure of the call processing.

In the case of wide-band digital transmission systems, fewer systems are required between nodes to carry the same traffic so that any failure affects a larger number of circuits.

The reduction in the number of switching centres (nodes) and the use of fewer inter-exchange transmission systems results in a network with reduced connectivity - in both the logical and physical senses. Thus, any failure of either a node or a transmission link, due to accident or human error, can have, to say the least, serious consequences for the network operator and the users so that the importance of an effective security (or survivability) plan cannot be over-emphasised.

8.7.3 Protection measures for conventional networks

In practice, survivability should be incorporated at the network planning stage-above all by providing as much route diversity as possible by closing loops in the tree-structure which usually represents the minimum-cost network. Other measures which should be taken at this stage, for conventional networks, include:

- load partitioning of switching centres, i.e. major nodes divided into two self-contained switching units with incoming and outgoing traffic divided between the two;
- increasing the network redundancy (capacity of nodes and links);
- <u>controlled</u> automatic alternative routing;
- implementation of one or more network management centres to monitor network performance and initiate corrective action, for example blocking traffic surges at their source;
- introducing loop configurations with traffic routed in both directions round the loop;
- providing alternative routes using other transmission media, for example radio-relay as an alternative to optical fibre cable.

A combination of some of the above measures will provide protection against the overload caused by network unavailability due to failures of nodes and links as well as the overload caused by traffic surges.

While route diversity is the most effective means of countering transmission failures it can be extremely expensive over certain types of terrain, as can the establishment of a ring structure over long distances. The only alternatives in this case are to use protection channels over the same physical route or establish a domestic satellite link.

8.7.4 Survivability of future networks

Survivability will, if possible, become even more important for the networks of the future, some examples of which are indicated in Chapter 5 (Target Networks). However, the introduction of networks using the synchronous digital hierarchy (SDH), asynchronous transfer mode (ATM) and other techniques will also provide further and better means of improving the survivability, and therefore the resilience to overloads and other phenomena. SDH as applied to local networks is briefly considered in Chapter 5 and Chapter 9 (paragraph 9.2).

Improvements to the robustness of the physical network alone are not sufficient to provide complete security and, as mentioned above, can be expensive and sometimes impractical over long distances. Security has therefore to be incorporated into the logical network (or ‘layer’) by the provision of some form of dynamic network reconfiguration by means of which the number of circuits, or the bandwidth, over specified sections or routes can be adjusted to accommodate unforeseen changes in demand.

In the case of ATM for example, which is at present the agreed target transfer mode for future broadband networks, the concept of virtual channels (VC), virtual paths (VP) and virtual path connections (VPC) is used in the routing philosophy. The concept of the virtual path is to provide logical direct routes, shared by several virtual channels, between switching nodes via intermediate cross-connect points. The VP is a semi-switched or permanent arrangement which provides the logical equivalent of a link between nodes -which are not necessarily directly connected by a physical link. Thus a distinction between the physical and logical network structures, and therefore the flexibility to re-arrange the logical structure according to the traffic requirements, is provided. This flexibility enables the network operator to make the best use of the available switching and transmission capacity under variable traffic conditions.

In an ATM network, the overflow routes provided for alternative routing need not involve further switching as they could be established as virtual paths, simply involving a greater number of sections. The network availability and security are enhanced by this ability to use a number of alternative paths.

As an extension of alternative routing, adaptive routing can be incorporated. In this case the second choice routes are not pre-determined but established according to current traffic conditions.

The concept of the virtual path has implications on the fundamental network structures. In the case of the subscriber access networks, cross-connect or add-drop points can be inserted between the (business) subscriber and the local exchange to enable connections to be made directly to different networks etc. thus changing the topology of the access network for example.

As far as the long-distance network is concerned, one study suggests the provision of VPs to every national local exchange with low traffic paths being routed over a transit network of cross-connect nodes and high traffic paths established as direct physical links. This is likely to give rise to a two-level switching hierarchy, the transit level being provided for international traffic and for routes for which a direct VP between local exchange areas is not justified, with a non-hierarchical network of cross-connect nodes which supports a logical network which can vary its structure according to the traffic conditions. The latter has obvious advantages for network security.

8.7.5 Some practical considerations

The effects of major node and transmission failures have been considered together with some of the means which are used to minimise these effects - for example the provision of a network configuration which gives ample scope for alternative routing, whether manual, adaptive or dynamic.

However, certain elementary preventive and implementation measures which could improve the inherent network reliability have also to be considered within the framework of security planning. The most obvious measure which should be taken in this respect is of course to ensure, by rigorous quality control, that all the network elements, from components to systems are of sufficiently high quality to be capable of reliable operation in their service environment.

Modern transmission equipment is inherently more reliable than in the past, due principally to its low power consumption. As mentioned in a later section (9.4.3) faults in this part of the network, mainly in isolated radio-relay stations, can usually be proved to be caused by failures in the primary power supplies. In principle, it is recommended that the use of continuously running generator sets should be avoided whenever possible and, if a public power supply cannot be connected, solar (photovoltaic) power should be used. Radio-relay equipment should not need an air-conditioned environment - in addition to excessive power requirements air-conditioning needs expert maintenance. In extreme climatic conditions passively conditioned accommodation should be used.

Outage times on transmission systems can be reduced by adopting the above measures in conjunction with preventive maintenance initiated by an efficient supervisory alarm system. In the case of remote stations, every effort should be made to provide all-weather access so that the preventive maintenance can be carried out in time.

Optical fibre cable systems will provide most of the capacity of long-distance networks in the future. When, as is usual, these cables are directly buried they are clearly vulnerable to damage by civil works. To some extent this vulnerability can be reduced by burying the cables in "open country", rather than alongside main roads for example, but it is preferable to seek more secure routes for these cables. Two possibilities are to run the cables along high (or low) voltage power lines or along railways. As mentioned in a later section, open-wire pole routes are available for this purpose in certain countries - and should be exploited. In any case provision for an alternative route, probably using radio-relay, has to be made for these high-capacity systems.

The security aspects of local networks using optical fibre junctions are considered in Chapter 9 (§9.2).