

International Telecommunication Union

ITU-T

G.8261/Y.1361

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

(04/2008)

SERIES G: TRANSMISSION SYSTEMS AND MEDIA,
DIGITAL SYSTEMS AND NETWORKS

Packet over Transport aspects – Quality and availability
targets

SERIES Y: GLOBAL INFORMATION
INFRASTRUCTURE, INTERNET PROTOCOL ASPECTS
AND NEXT-GENERATION NETWORKS

Internet protocol aspects – Transport

**Timing and synchronization aspects in packet
networks**

Recommendation ITU-T G.8261/Y.1361



ITU-T G-SERIES RECOMMENDATIONS
TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

INTERNATIONAL TELEPHONE CONNECTIONS AND CIRCUITS	G.100–G.199
GENERAL CHARACTERISTICS COMMON TO ALL ANALOGUE CARRIER-TRANSMISSION SYSTEMS	G.200–G.299
INDIVIDUAL CHARACTERISTICS OF INTERNATIONAL CARRIER TELEPHONE SYSTEMS ON METALLIC LINES	G.300–G.399
GENERAL CHARACTERISTICS OF INTERNATIONAL CARRIER TELEPHONE SYSTEMS ON RADIO-RELAY OR SATELLITE LINKS AND INTERCONNECTION WITH METALLIC LINES	G.400–G.449
COORDINATION OF RADIOTELEPHONY AND LINE TELEPHONY	G.450–G.499
TRANSMISSION MEDIA AND OPTICAL SYSTEMS CHARACTERISTICS	G.600–G.699
DIGITAL TERMINAL EQUIPMENTS	G.700–G.799
DIGITAL NETWORKS	G.800–G.899
DIGITAL SECTIONS AND DIGITAL LINE SYSTEM	G.900–G.999
QUALITY OF SERVICE AND PERFORMANCE – GENERIC AND USER-RELATED ASPECTS	G.1000–G.1999
TRANSMISSION MEDIA CHARACTERISTICS	G.6000–G.6999
DATA OVER TRANSPORT – GENERIC ASPECTS	G.7000–G.7999
PACKET OVER TRANSPORT ASPECTS	G.8000–G.8999
Ethernet over Transport aspects	G.8000–G.8099
MPLS over Transport aspects	G.8100–G.8199
Quality and availability targets	G.8200–G.8299
Service Management	G.8600–G.8699
ACCESS NETWORKS	G.9000–G.9999

For further details, please refer to the list of ITU-T Recommendations.

Recommendation ITU-T G.8261/Y.1361

Timing and synchronization aspects in packet networks

Summary

Recommendation ITU-T G.8261/Y.1361 defines synchronization aspects in packet networks. It specifies the maximum network limits of jitter and wander that shall not be exceeded. It specifies the minimum equipment tolerance to jitter and wander that shall be provided at the boundary of these packet networks at TDM and synchronization interfaces. It also outlines the minimum requirements for the synchronization function of network elements.

The requirements for the jitter and wander characteristics that are specified in this Recommendation must be adhered to in order to ensure interoperability of equipment produced by different manufacturers and a satisfactory network performance.

Source

Recommendation ITU-T G.8261/Y.1361 was approved on 29 April 2008 by ITU-T Study Group 15 (2005-2008) under Recommendation ITU-T A.8 procedure.

Keywords

Clock, jitter, synchronization, wander.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications, information and communication technologies (ICTs). The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g. interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, ITU had received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at <http://www.itu.int/ITU-T/ipr/>.

© ITU 2009

All rights reserved. No part of this publication may be reproduced, by any means whatsoever, without the prior written permission of ITU.

CONTENTS

	Page
1 Scope	1
2 References.....	1
3 Definitions	3
3.1 Terms defined elsewhere.....	3
3.2 Terms defined in this Recommendation.....	3
4 Abbreviations.....	5
5 Conventions	6
6 General.....	6
6.1 Packet network synchronization requirements.....	7
6.2 TDM timing requirements.....	8
6.3 Synchronization network engineering in packet networks.....	8
6.4 Timing requirements at edge versus timing requirements in core networks..	9
6.5 PNT domain and CES domain	9
7 Reference timing signal distribution over packet networks (PNT domain)	9
7.1 Plesiochronous and network synchronous methods.....	9
7.2 Packet-based methods	11
8 Timing recovery for constant bit rate services transported over packet networks (CES domain)	12
8.1 Network synchronous operation.....	12
8.2 Differential methods.....	13
8.3 Adaptive methods.....	14
8.4 Reference clock available at the TDM end systems.....	14
9 Network limits	15
9.1 CES network limits.....	15
9.2 PNT network limits	20
10 Impact of impairments in the packet network on timing distribution and service clock recovery.....	25
10.1 Packet transfer delay and delay variation.....	26
10.2 Impacts from packet impairments	31
11 Impact of the reference clock impairment on the service clock recovery.....	32
11.1 Impairments for the network synchronous operation methods.....	32
11.2 Impairments for the differential method.....	33
12 Results and consequences of the different synchronization methods over packet network reference models.....	34
12.1 CES domain recommendations	34
12.2 PNT domain recommendations	36
Annex A – Proposed network architecture for synchronous Ethernet.....	38
A.1 PRC Location	38

	Page
A.2 Limiting jitter and wander of synchronous Ethernet.....	38
A.3 Considerations on the design of synchronization network based on synchronous Ethernet	39
A.4 Example of timing distribution via synchronous Ethernet.....	40
A.5 Interworking of Ethernet and synchronous Ethernet interfaces	40
Annex B – IWF functional partitioning into CES and PNT IWF and network examples.....	44
B.1 General	44
B.2 IWF clocks.....	45
B.3 Network examples	48
Annex C – CES IWF synchronization related requirements	51
C.1 Traffic interfaces	51
C.2 Synchronization interfaces	51
C.3 IWF synchronization function.....	52
Annex D – Network applications and requirements for clocks specified in G.8262/Y.1362..	53
Appendix I – Characteristics of Ethernet switches and networks.....	54
I.1 Delay characteristics of Ethernet switches	54
I.2 Characteristics of switched Ethernet networks.....	57
Appendix II – Stabilization period.....	59
Appendix III – Considerations on packet-based methods	60
Appendix IV – Applications and use cases.....	61
IV.1 Background.....	61
IV.2 Wireless	61
IV.3 Infrastructure	63
IV.4 Media gateway.....	63
Appendix V – Packet networks reference models	64
V.1 Ethernet networks models	64
V.2 Other network models	67
Appendix VI – Measurement guidelines for packet-based methods	71
VI.1 Measurement reference points.....	71
VI.2 Input traffic characteristics	72
VI.3 Test topologies for adaptive methods.....	73
VI.4 Test Topologies for differential methods	80
VI.5 Test for two-way protocols.....	81
Appendix VII – Wander limits in Deployment Case 1	88
VII.1 Limits for the 2048 kbit/s interface	88
VII.2 Limits for the 1544 kbit/s interface	88
Appendix VIII – Synchronization status messaging in synchronous Ethernet PHY	89
Appendix IX – IWF examples	90

	Page
Appendix X – Considerations on measurement of synchronous Ethernet according to ITU-T methodologies in comparison with IEEE jitter measurements	93
Appendix XI – Relationship between requirements contained in this Recommendation and other key synchronization related Recommendations	94
Appendix XII – Basic principles of timing over packet networks.....	96
XII.1 General	96
XII.2 Packet delay variation mitigation by packet selection	99
XII.3 Comparison of packet-based and synchronous PHY methods.....	99
XII.4 Existing standards.....	100
Bibliography.....	101

Recommendation ITU-T G.8261/Y.1361

Timing and synchronization aspects in packet networks

1 Scope

This Recommendation defines synchronization aspects in packet networks. It specifies the maximum network limits of jitter and wander that shall not be exceeded. It specifies the minimum equipment tolerance to jitter and wander that shall be provided at the boundary of these packet networks at TDM and synchronization interfaces. It also outlines the minimum requirements for the synchronization function of network elements.

In particular, two main issues are addressed in this Recommendation: the distribution of a synchronization network clock signal over a packet network (PNT domain), and the distribution of a service clock signal over a packet network (CES domain).

NOTE – The application of the transport of SDH signals over packet networks is only partly covered and some aspects are for further study.

The packet networks that are in the scope of this Recommendation are currently limited to the following scenarios:

- Ethernet [IEEE 802.3], [IEEE 802.1D], [IEEE 802.1ad], [IEEE 802.1Q] and [b-IEEE P802.1Qay]
- MPLS [IETF RFC 3031] and [ITU-T G.8110]
- T-MPLS [ITU-T G.8110.1]
- IP [IETF RFC 791] and [IETF RFC 2460]

The physical layer that is relevant to this Recommendation is the Ethernet media types as defined in [IEEE 802.3]. Other physical layers can be relevant and may be addressed in a future version of this Recommendation.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

- [ITU-T G.691] Recommendation ITU-T G.691 (2006), *Optical interfaces for single channel STM-64 and other SDH systems with optical amplifiers.*
- [ITU-T G.702] Recommendation ITU-T G.702 (1988), *Digital hierarchy bit rates.*
- [ITU-T G.703] Recommendation ITU-T G.703 (2001), *Physical/electrical characteristics of hierarchical digital interfaces.*
- [ITU-T G.705] Recommendation ITU-T G.705 (2000), *Characteristics of plesiochronous digital hierarchy (PDH) equipment functional blocks.*
- [ITU-T G.803] Recommendation ITU-T G.803 (2000), *Architecture of transport networks based on the synchronous digital hierarchy (SDH).*
- [ITU-T G.811] Recommendation ITU-T G.811 (1997), *Timing characteristics of primary reference clocks.*

- [ITU-T G.812] Recommendation ITU-T G.812 (2004), *Timing requirements of slave clocks suitable for use as node clocks in synchronization networks.*
- [ITU-T G.813] Recommendation ITU-T G.813 (2003), *Timing characteristics of SDH equipment slave clocks (SEC).*
- [ITU-T G.822] Recommendation ITU-T G.822 (1988), *Controlled slip rate objectives on an international digital connection.*
- [ITU-T G.823] Recommendation ITU-T G.823 (2000), *The control of jitter and wander within digital networks which are based on the 2048 kbit/s hierarchy.*
- [ITU-T G.824] Recommendation ITU-T G.824 (2000), *The control of jitter and wander within digital networks which are based on the 1544 kbit/s hierarchy.*
- [ITU-T G.825] Recommendation ITU-T G.825 (2000), *The control of jitter and wander within digital networks which are based on the synchronous digital hierarchy (SDH).*
- [ITU-T G.957] Recommendation ITU-T G.957 (2006), *Optical interfaces for equipments and systems relating to the synchronous digital hierarchy.*
- [ITU-T G.959.1] Recommendation ITU-T G.959.1 (2008), *Optical transport network physical layer interfaces.*
- [ITU-T G.8010] Recommendation ITU-T G.8010/Y.1306 (2004), *Architecture of Ethernet layer networks.*
- [ITU-T G.8110] Recommendation ITU-T G.8110/Y.1370 (2005), *MPLS layer network architecture.*
- [ITU-T G.8110.1] Recommendation ITU-T G.8110.1/Y.1370.1 (2006), *Architecture of Transport MPLS (T-MPLS) layer network.*
- [ITU-T G.8262] Recommendation ITU-T G.8262/Y.1362 (2007), *Timing characteristics of synchronous Ethernet equipment slave clock (EEC).*
- [ITU-T G.8264] Recommendation ITU-T G.8264/Y.1364 (2008), *Timing distribution through packet networks.*
- [ITU-T O.171] Recommendation ITU-T O.171 (1997), *Timing jitter and wander measuring equipment for digital systems which are based on the plesiochronous digital hierarchy (PDH).*
- [ITU-T O.172] Recommendation ITU-T O.172 (2005), *Jitter and wander measuring equipment for digital systems which are based on the synchronous digital hierarchy (SDH).*
- [ITU-T Y.1411] Recommendation ITU-T Y.1411 (2003), *ATM-MPLS network interworking – Cell mode user plane interworking.*
- [ITU-T Y.1540] Recommendation ITU-T Y.1540 (2002), *Internet protocol data communication service – IP packet transfer and availability performance parameters.*
- [ITU-T Y.1561] Recommendation ITU-T Y.1561 (2004), *Performance and availability parameters for MPLS networks.*
- [ITU-T Y.1731] Recommendation ITU-T Y.1731 (2006), *OAM functions and mechanisms for Ethernet based networks.*
- [IEEE 802] IEEE 802-2001, *IEEE standard for local and metropolitan area networks: Overview and architecture* <<http://standards.ieee.org/getieee802/802.html>>.

- [IEEE 802.1D] IEEE 802.1D-2004, *IEEE Standard for local and metropolitan area networks: Media Access Control (MAC) Bridges*
<<http://standards.ieee.org/getieee802/download/802.1D-2004.pdf>>.
- [IEEE 802.1Q] IEEE 802.1Q-2005, *IEEE Standard for local and metropolitan area networks: Virtual bridged local area networks*
<<http://standards.ieee.org/getieee802/download/802.1Q-2005.pdf>>.
- [IEEE 802.1ad] IEEE 802.1ad-2005, *IEEE Standard for local and metropolitan area networks: Virtual bridged local area networks – Amendment 4: Provider Bridges*
<<http://standards.ieee.org/getieee802/download/802.1ad-2005.pdf>>.
- [IEEE 802.3] IEEE 802.3-2005, *Part 3: Carrier sense multiple access with collision detection (CSMA/CD) access method and physical layer specifications*
<<http://standards.ieee.org/getieee802/802.3.html>>.
- [IETF RFC 791] IETF RFC 791 (1981), *Internet Protocol (IP)*
<<http://www.ietf.org/rfc/rfc0791.txt?number=791>>.
- [IETF RFC 2460] IETF RFC 2460 (1998), *Internet Protocol, Version 6 (IPv6) Specification*
<<http://www.ietf.org/rfc/rfc2460.txt?number=2460>>.
- [IETF RFC 3031] IETF RFC 3031 (2001), *Multiprotocol Label Switching Architecture*
<<http://www.ietf.org/rfc/rfc3031.txt?number=3031>>.

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following terms defined elsewhere:

3.1.1 asynchronous interface: See [ITU-T G.823].

3.1.2 interworking function (IWF): See [ITU-T Y.1411]. More details and examples are shown in Annex B and Appendix IX.

3.1.3 synchronous interface: See [ITU-T G.823].

3.1.4 traffic interface: See [ITU-T G.823].

3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1 adaptive clock recovery: Clock recovery technique that does not require the support of a network-wide synchronization signal to regenerate the timing. In this case, the timing recovery process is based on the (inter-)arrival time of the packets (e.g., timestamps or CES packets). The information carried by the packets could be used to support this operation. Two-way or one-way protocols can be used.

NOTE – For the purpose of this Recommendation, this definition relates to frequency synchronization only.

3.2.2 CES IWF: The CES IWF is the set of functions within the IWF that supports the service clock domain (see Figure B.3). This includes the function to recover the service clock timing.

3.2.3 circuit emulation services (CES) island: Segment of a network, based on packet-switched technologies, that emulates either the characteristics of a circuit-switched network or of a PDH/SDH transport network, in order to carry CBR services (e.g., E1).

3.2.4 frequency source traceability: Frequency source traceability is a relationship where the frequency of all clocks in a system is referenced back to a single physical clock. Under normal operating conditions, all clocks will have the same average frequency in a source-traceable system.

Thus, the phase error or maximum time interval error (MTIE) between all clocks in such a system is bounded.

NOTE – A different case is when the clocks have frequency traceability towards accurate master clocks (not necessarily the same pieces of equipment). This is connected to the concept of plesiochronous as defined in [ITU-T G.810]. An example is when the clocks have the "PRC traceability" (i.e., traceability to ITU-T G.811 clocks) in a synchronization network based on the distributed PRC architecture.

3.2.5 network clock: The clock generating the network clock signal.

3.2.6 network clock domain: Set of functions dedicated to support the synchronization network (network clock).

3.2.7 network clock signal: A reference timing signal that is used as a reference to allow mapping and demapping of a service clock at ingress and egress points of the network respectively. In some applications, the signal could be asynchronous and generated by free running clocks with low requirements in terms of frequency accuracy (e.g., in the Ethernet network, the physical layer can operate up to ± 100 ppm). In other applications, an accurate reference timing signal is needed. In this case, the signal is typically traceable to a PRC under normal conditions, and the distribution of this signal across the network is accomplished by a synchronization network.

NOTE – For the purpose of this Recommendation, it is assumed that a properly high accurate signal is always involved. Due to that, the network clock signal definition can be considered to coincide with the definition of synchronization network clock signal, and the two terms are used interchangeably throughout this Recommendation.

3.2.8 network-synchronous operation: Synchronization of the physical layer (usually by a timing distribution of a timing signal traceable to a primary reference clock (PRC), see [ITU-T G.811]).

3.2.9 Packet network timing function (PNT-F): The set of functions within the IWF that supports the synchronization network clock domain (see Figure B.2). This includes the function to recover and distribute the timing carried by the synchronization network. The PNT-F clocks may be part of the IWF or may be part of any other network element in the packet network.

When the PNT-Fs are part of the IWF, they may support the CES IWF and/or change the layer over which timing is carried (i.e., from packet to physical layer and vice versa).

3.2.10 service clock: The clock generating the service clock signal.

3.2.11 service clock domain: Set of functions dedicated to support the CES timing function (service clock).

3.2.12 service clock signal: The timing information that is associated with a specific service supported by a network. For instance, in case of E1 TDM service, the timing shall be 2048 kbit/s ± 50 ppm.

3.2.13 synchronization network clock: The equipment that provides the timing signal in the synchronization network.

3.2.14 synchronization network clock signal: The reference timing signal distributed by the synchronization network. This signal is traceable to an accurate master (e.g., PRC).

3.2.15 time division multiplex (TDM): A term that conventionally refers to the isochronous bit streams used in telephony networks; in particular, those belonging to PDH (plesiochronous digital hierarchy) as described in [ITU-T G.705]. The bit rates traditionally used in various regions of the world are detailed in [ITU-T G.702]. Examples of the signals covered by the TDM definition are those belonging to PDH and SDH hierarchies.

3.2.16 stabilization period: The period beginning at the point in time when a validated timing source has been selected by the IWF and ending when the output timing characteristics are within the output jitter and wander requirements.

3.2.17 wander budget (of a network island): Wander generated at the output of a network island when an ideal reference timing signal is the input at the first network element of this network island.

4 Abbreviations

This Recommendation uses the following abbreviations:

3GPP	Third Generation Partnership Project
ATM	Asynchronous Transfer Mode
BS	Base Station
CBR	Constant Bit Rate
CDMA	Code Division Multiple Access
CE	Customer Equipment
CES	Circuit Emulation Service
DUT	Device Under Test
EEC	Synchronous Ethernet Equipment Clock
ESMC	Ethernet Synchronization Messaging Channel
FDD	Frequency Division Duplex
FE	Fast Ethernet
GE	Gigabit Ethernet
GPS	Global Positioning System
GSM	Global System for Mobile communications
IP	Internet Protocol
IP DSLAM	IP Digital Subscriber Line Access Multiplexer
IWF	Interworking Function
MAC	Medium Access Control
M-CMTS	Modular Cable Modem Termination System
METROE	METRO Ethernet
MPEG	Moving Picture Experts Group
MRTIE	Maximum Relative Time Interval Error
MSAN	MultiService Access Node
MTIE	Maximum Time Interval Error
NTP	Network Time Protocol
OLT	Optical Line Termination
OTN	Optical Transport Network
PDH	Plesiochronous Digital Hierarchy
PDV	Packet Delay Variation
PEC	Packet-based Equipment Clock
PHY	PHYsical (layer)
PNT	Packet Network Timing

PNT-F	PNT-Function
PRC	Primary Reference Clock
PSC-A	Packet-based Service Clock-Adaptive
PSC-D	Packet-based Service Clock-Differential
PSTN	Public Switched Telephone Network
PTP	Precision Time Protocol
QL	Quality Level
SASE	Stand Alone Synchronization Equipment
SDH	Synchronous Digital Hierarchy
SEC	SDH Equipment Clock
SLA	Service Level Agreement
SNTP	Simple Network Time Protocol
SRTS	Synchronous Residual Time Stamp
SSM	Synchronization Status Message
SSU	Synchronization Supply Unit
STM	Synchronous Transfer Mode
TCP	Transmission Control Protocol
TDD	Time Division Duplex
TDEV	Time DEVIation
TDM PW	TDM PseudoWire
TDM	Time Division Multiplex
ToD	Time of Day
UI	Unit Interval
UTC	Coordinated Universal Time
WCDMA	Wideband Code Division Multiple Access

5 Conventions

The terms "packets" and "frames" are used interchangeably throughout this Recommendation.

Within this Recommendation, the term "Ethernet" refers to an interface as defined in [IEEE 802.3] and that does not comply with the additional timing requirements of synchronous Ethernet as specified in this Recommendation, in [ITU-T G.8262] and in [ITU-T G.8264].

6 General

Packet switching was originally introduced to handle asynchronous data.

However, for new applications, such as the transport of TDM service and the distribution of synchronization over packet networks, the strict synchronization requirements of those applications must be considered.

The ongoing evolution in telecommunications increases the likelihood of hybrid packet/circuit environments for voice and voice band data services. These environments combine packet

technologies (e.g., ATM, IP, Ethernet) with traditional TDM systems. Under these conditions, it is critical to ensure that an acceptable level of quality is maintained (e.g., limited slip rate).

Synchronization in TDM networks is well understood and implemented. Typically, a TDM circuit service provider will maintain a timing distribution network, providing synchronization traceable to a primary reference clock (i.e., clock compliance with [ITU-T G.811]).

The timing and synchronization aspects addressed in this Recommendation are initially concerned with networks with physical layer based on Ethernet media types as defined in [IEEE 802.3] (see Scope, clause 1).

Functional architecture for Ethernet networks are defined in [ITU-T G.8010].

In the context of this Recommendation, the highest layers (e.g., layer 7 in OSI model) refer to applications transported over the packet networks. Real-time applications have relatively tight timing requirements concerning delay and delay variation. Some applications might resolve their timing issues within higher layers (e.g., MPEG-2); other applications rely on the timing support provided by one or more of the lower layers (e.g., physical layer).

This Recommendation aims at describing different methods to obtain the synchronization-related requirements. Both CES and PNT domains are considered, and different requirements are described.

Additionally, the requirements for interfaces and equipments that are part of the Ethernet and synchronous Ethernet network are described. It also gives recommendations when to apply different types of synchronization methods.

Some considerations on applicable synchronization requirements in a packet-based network are summarized in the following clauses.

This Recommendation primarily deals with CES in public network environments. In some private network applications involving circuit emulation, it may be sufficient to distribute a non-PRC quality level common clock towards CES IWF nodes. However, the use of synchronization timing below PRC quality level could result in interworking difficulties between different network domains, such as interconnection involving multiple public networks providers.

The use of a non-PRC quality level common clock is for further study.

6.1 Packet network synchronization requirements

The nodes involved in packet oriented transmission technology (e.g., ATM network nodes) do not require any synchronization for the implementation of the packet switching function. In fact, at any entrance point of a packet switch, an individual device shall provide packet timing adaptation (for instance, cell timing adaptation in case of ATM switch) of the incoming signal to the internal timing. For instance, in the case of ATM networks, the principle to cater for frequency differences is to use idle cell stuffing. Transmission links therefore, in principle, need not to be synchronized with each other.

However, as the packet network evolves to integrate TDM-based applications, i.e., when transporting a CBR stream over a packet network and when interworking with PSTN networks, the packet network shall provide correct timing at the traffic interfaces.

This means that the requirements on synchronization functions in packet networks, especially on the boundary of the packet networks, are dependent on the services carried over the network. For TDM based services, the IWF may require network-synchronous operation in order to provide acceptable performance.

6.2 TDM timing requirements

The transport of TDM signals through packet networks requires that the signals at the output of the packet network comply with TDM timing requirements, this is crucial to enable interworking with TDM equipment.

These requirements are independent of the type of information (voice or data) transported by the TDM signal.

The adaptation of TDM signals into the packet network is called circuit emulation services (CESs).

The timing requirements that are applicable are: jitter and wander limits at traffic and/or synchronization interfaces, long-term frequency accuracy (which can influence the slip performance), and total delay (which is critical for real-time services, for instance voice service).

6.2.1 PDH timing requirements

The PDH timing requirements for traffic interfaces are mainly related to jitter, wander and slip performance.

At the input of the network element at the boundary of a packet network, jitter and wander tolerance requirements apply. At the output of the network element at the egress of the packet network, jitter and wander generation requirements apply.

These values are specified in [ITU-T G.823] for the network based on 2048 kbit/s hierarchy and [ITU-T G.824] for the network based on 1544 kbit/s hierarchy.

In addition, [ITU-T G.822] specifies the applicable slip rate objectives. This is the case when the clock of the equipment that generates the TDM signal and the clock used in the equipment recovering the TDM signal from the packets are different and the slip buffer is needed in the application.

6.2.2 Synchronization interfaces requirements

In the case where the PDH signals are defined as synchronization interfaces, the synchronization requirements are more stringent than for 2048 kbit/s and 1544 kbit/s traffic interfaces. The synchronization interface requirements for PDH interfaces are also defined in [ITU-T G.823] and [ITU-T G.824].

6.2.3 SDH timing requirements

Any STM-N signal must be compliant with [ITU-T G.825]. The relevant requirements refer to the jitter and wander tolerance applicable at the input of the network element at the boundary of a packet network that receives the STM-N data, and jitter and wander generation applicable at the output of the network element generating STM-N traffic at the other end of the packet network.

In the case of STM-N signals, there are no distinctions between traffic and synchronization interfaces as all STM-N signals are defined as synchronization interfaces.

6.3 Synchronization network engineering in packet networks

The driving force for much of this work is to cater for the synchronization needs of the application or, in general, the need of certain technologies (for instance base stations in GSM and WCDMA networks). In order to achieve such a goal, operators have therefore to distribute a reference timing signal of suitable quality to the network elements processing the application.

One approach is to follow a distributed PRC strategy (for instance by means of GPS technologies). An alternative approach is based on a master-slave strategy. The engineering rules for designing the synchronization network in these cases are well understood and documented (see for example [ITU-T G.803]) when the underlying transport of the packets (e.g., Ethernet frames) will run over the existing synchronous technologies (PDH or SDH networks). On the other hand, when the underlying

transport is based on non-synchronous technologies (i.e., Ethernet), alternative approaches shall be considered. This will be further analysed in clause 7.

6.4 Timing requirements at edge versus timing requirements in core networks

Different performances can be requested in case the packet network is part of an access network or is the underlying layer of the core network.

The distribution of a synchronization reference over a portion of a core network may be requested to comply with strict jitter and wander requirements (i.e., [ITU-T G.823], [ITU-T G.824] for synchronization interfaces and [ITU-T G.825]).

On the other hand, in the access network, requirements may be relaxed to allow a distribution of a timing reference signal with performance sufficient (e.g., lower than PRC quality level) to support the timing requirements of the end node (for instance, a base station, or a V.90 modem). More information is provided in Appendix IV.

6.5 PNT domain and CES domain

This Recommendation addresses two main different issues:

- 1) How to carry a synchronization network clock signal over a packet network:
 - This issue is related to the PNT domain, and refers to the network clock (see definitions).
 - Guidance regarding this issue is provided in clause 7.
- 2) How to carry a service clock signal:
 - This issue is related to the CES domain, and refers to the service clock (see definitions).
 - Guidance regarding this issue is provided in clause 8.

Further information regarding PNT and CES domains are provided in Annex B.

7 Reference timing signal distribution over packet networks (PNT domain)

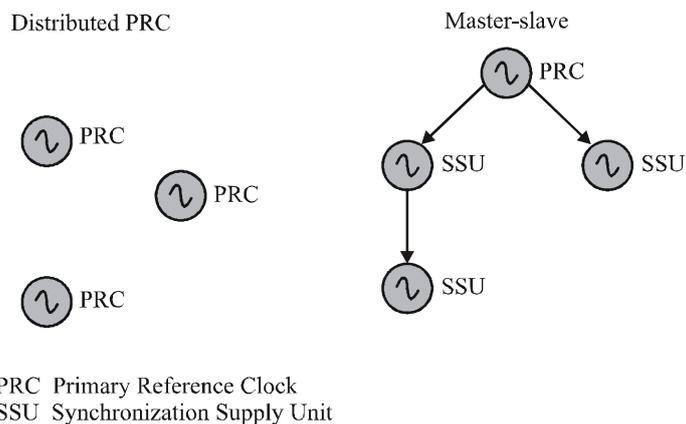
In order to fulfil the applicable synchronization requirements, it should be possible to distribute a reference timing signal with proper phase stability and frequency accuracy characteristics.

Two main classes of methods are identified in this Recommendation:

- Plesiochronous and network synchronous methods (that is, reference timing signal distributed over the synchronous physical layer),
- Packet-based methods.

7.1 Plesiochronous and network synchronous methods

The first class of methods refers to PRC distributed method (for instance, based on GPS), or master-slave method using a synchronous physical layer (e.g., STM-N), see Figure 1. These methods are widely implemented to synchronize the TDM networks.



G.8261-Y.1361(08)_F01

Figure 1 – Distributed PRC and master slave methods

Ethernet networks are free-running (± 100 ppm). However, in case of synchronous Ethernet, it is possible to design a master-slave synchronization architecture at the physical layer. In this case, the physical layer can be used to provide reference timing signal distribution over packet networks, from backbone level to access level. This method can also be used to provide timing recovery at the IWFs for CBR services transported over packet networks (network synchronous operations). It could also be used to provide a reference timing signal down to edge access equipment in a pure Ethernet network supporting synchronous Ethernet.

Clause 7.1.1 details a high-level method of achieving a synchronous Ethernet network.

7.1.1 Synchronous Ethernet networks

The general concept of delivering a physical layer clock from the Ethernet switch over the synchronous Ethernet is given in Figure 2.

A reference timing signal traceable to a PRC is injected into the Ethernet switch using an external clock port. This signal is extracted and processed via a synchronization function before injecting timing onto the Ethernet bit stream. The synchronization function provides filtering and may require holdover. The clock supporting synchronous Ethernet networks is called EEC, synchronous Ethernet equipment clock (see [ITU-T G.8262]).

As shown in the figure, there may be a number of Ethernet switches involved in the distribution of the reference timing signal. In such cases, the synchronization function within these Ethernet switches must be able to recover synchronization "line timing" from the incoming bit stream.

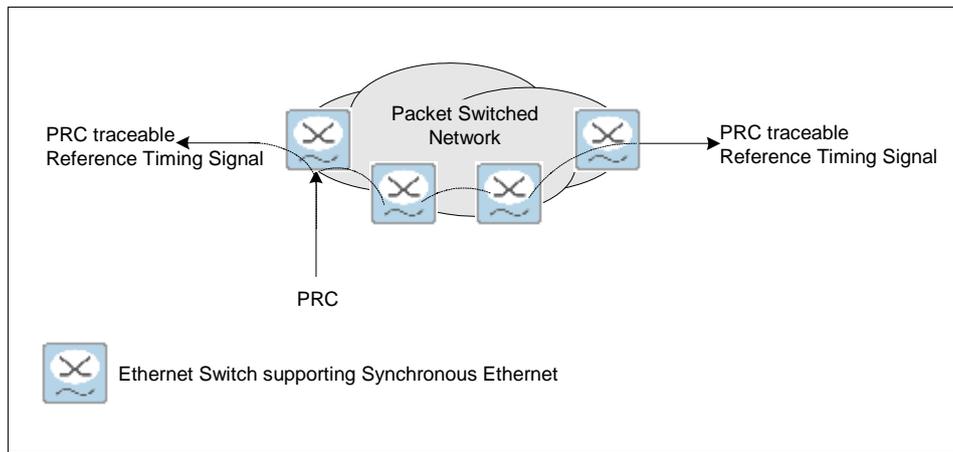


Figure 2 – Example of master slave synchronization network over synchronous Ethernet

As part of the architecture, a distinction should be made between the network clock and the service clock as described below.

The term synchronous Ethernet applies to the network clock, that is the clock that is controlling the bit rate leaving the Ethernet switch. This clock shall comply with [ITU-T G.8262].

Within existing Ethernet technology, the service is effectively asynchronous. In synchronous Ethernet, existing Ethernet services will continue to be mapped into and out of the Ethernet physical layer at the appropriate rates as generated by the service clocks.

A proposed architecture for the synchronization networks based on synchronous Ethernet is described in Annex A.

NOTE – The synchronous Ethernet equipments have to comply with [b-ITU-T G.781] that specifies the synchronization layer, and [ITU-T G.8264] that specifies the SSM for synchronous Ethernet.

7.2 Packet-based methods

The second class of methods relies on timing information carried by the packets. In this case, the timing could be carried by dedicated time stamp messages as shown in Figure 3. In case the physical layer is not synchronous, this is the only alternative to a PRC distributed approach. The principles underlying such methods are outlined in Appendix XII.

The time stamp can be based on several protocols. Examples of protocols are NTP and PTP (see clause XII.4).

The PTP protocol uses time stamps for synchronizing clocks in the network in a master-slave hierarchy. It can be used to distribute frequency and/or time of day (ToD) information. PTP was originally introduced for Industrial Automation and Test and Measurement Industries; however, a new version (see clause XII.4) has included several updates to allow it to be used for Telecom.

NTP and SNTP are protocols traditionally used to distribute time of day information. The same packets may be used to distribute frequency information as well.

Packet-based methods are adaptive in nature, since they do not require the support of a network-wide synchronization reference. Therefore the performance is impacted by packet delay variation in the network (see clause 10). In order to minimize the impact from the packet network using either PTP or NTP packets, specific algorithms may need to be implemented at the client side, depending on the required accuracy (see Appendices III and IV).

Additional requirements on the intermediate nodes in the network can be considered to enhance the performance of these methods. It should be noted that, especially in cases where legacy equipment is used, this may not always be feasible.

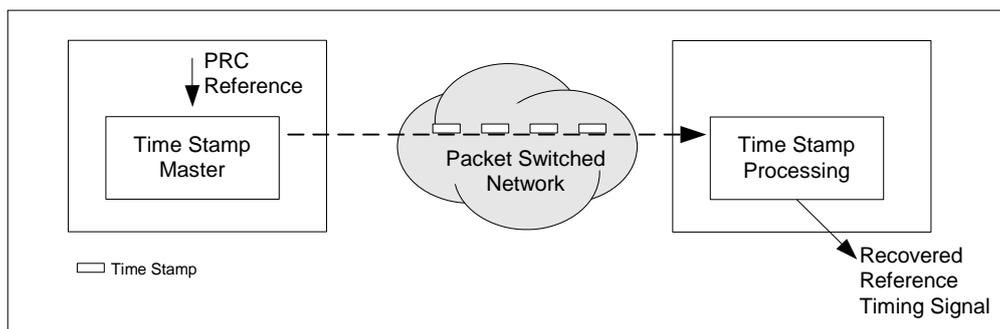


Figure 3 – Example of packet-based methods with timing distribution of the reference timing signal via time stamps

NOTE – Performance and details of packet-based methods are under study. For example, the various PTP options and parameter values appropriate for telecommunication applications are being documented as part of the so-called "Telecom Profile(s)", which is currently under development.

Some initial information and related Recommendations on packet-based methods are provided in clause 12.2.2.

The clock supporting packet-based methods is called packet-based equipment clock (PEC), (see Annex B).

8 Timing recovery for constant bit rate services transported over packet networks (CES domain)

CBR (Constant Bit Rate) services (e.g., circuit emulated TDM signal) require that the timing of the signal is similar on both ends of the packet network and is handled by the IWF responsible for delivering the constant bit rate stream. The notion of service clock preservation is that the incoming service clock frequency be replicated as the outgoing service clock frequency when considered in terms of a long-term average. It does not imply that wander on the incoming TDM signal be replicated on the outgoing TDM signal.

The four operating methods identified within this Recommendation are described in the following clauses:

- Network synchronous operation
- Differential methods
- Adaptive methods
- Reference clock available at the TDM end systems

8.1 Network synchronous operation

This method refers to the fully network-synchronous operation by using a PRC traceable network derived clock or a local PRC (e.g., GPS) as the service clock (see Figure 4). This implies the availability of a PRC reference. It should be highlighted that this method does not preserve the service timing.

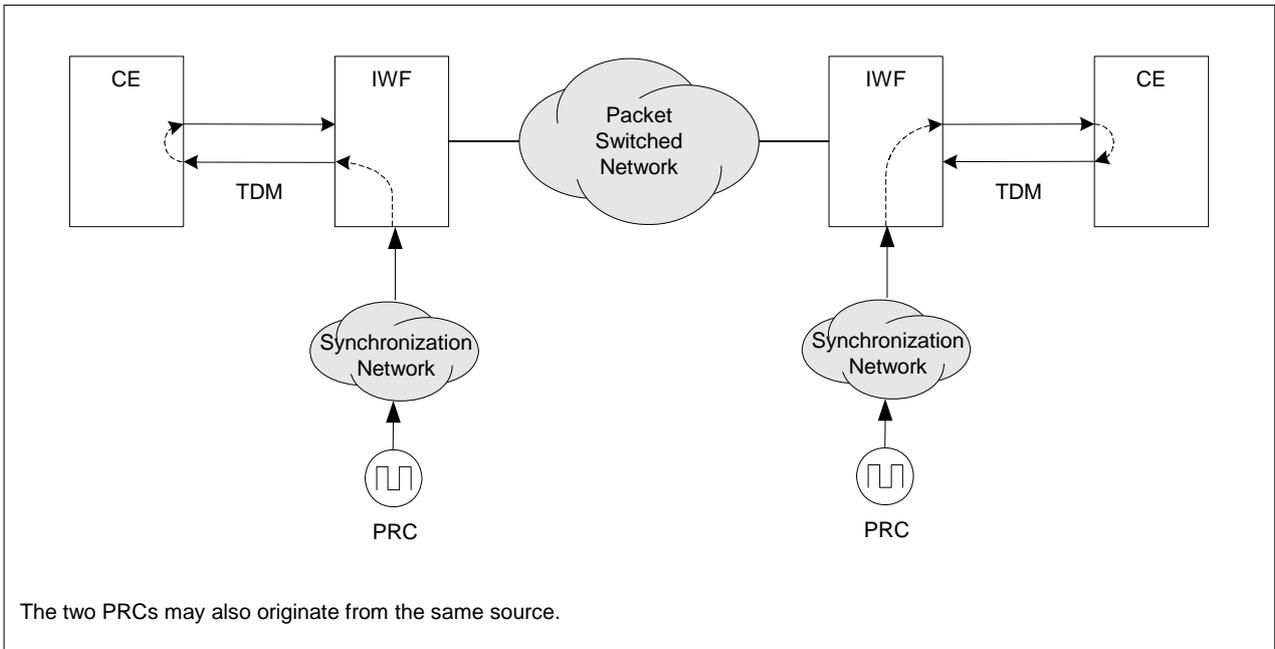


Figure 4 – Example of network synchronous operation

NOTE – The reference timing signal at the input to the IWF shall comply with synchronization interfaces as defined in [ITU-T G.823] and [ITU-T G.824].

8.2 Differential methods

According to the differential methods, the difference between the service clock and the reference clock is encoded and transmitted across the packet network (Figure 5). The service clock is recovered on the far end of the packet network making use of a common reference clock. The synchronous residual time stamp (SRTS) method [b-ITU-T I.363.1] is an example of this family of methods. It should be highlighted that the method can preserve the service timing.

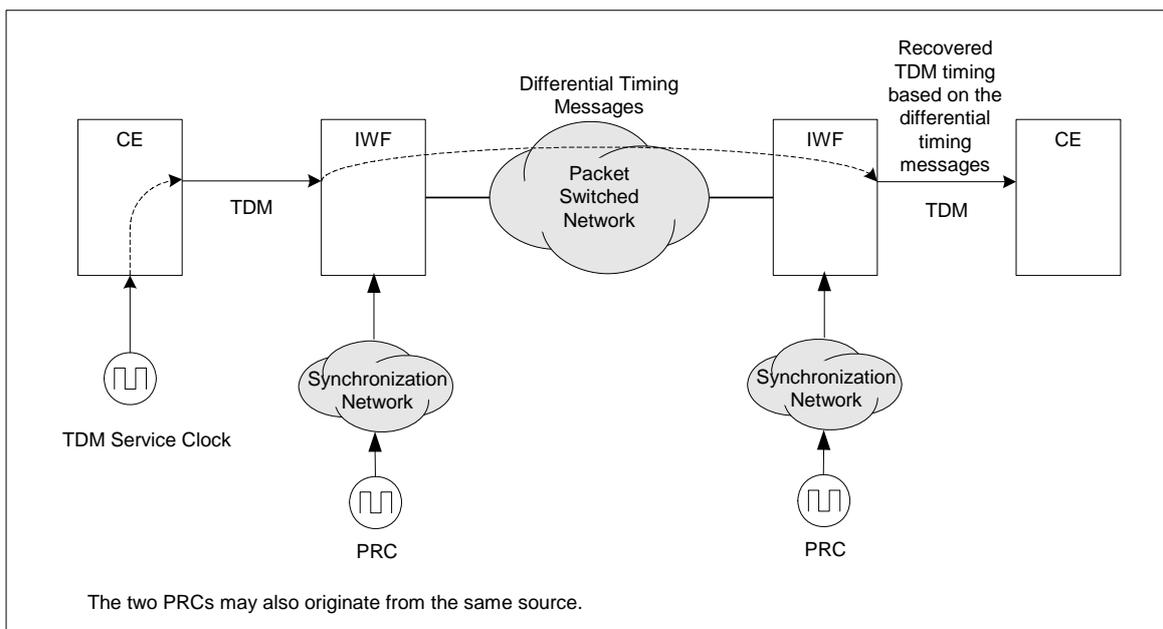


Figure 5 – Example of timing recovery operation based on differential methods

NOTE 1 – Differential methods may work with IWF reference clocks that are not PRC traceable. The use of non-PRC traceable clocks is application-dependent and is not within the scope of this Recommendation.

NOTE 2 – The reference timing signal at the input of the IWF shall comply with synchronization interfaces as defined in [ITU-T G.823] and [ITU-T G.824].

8.3 Adaptive methods

In the adaptive method, the timing can be recovered based on the inter-arrival time of the packets or on the fill level of the jitter buffer. It should be highlighted that the method preserves the service timing (see Figure 6).

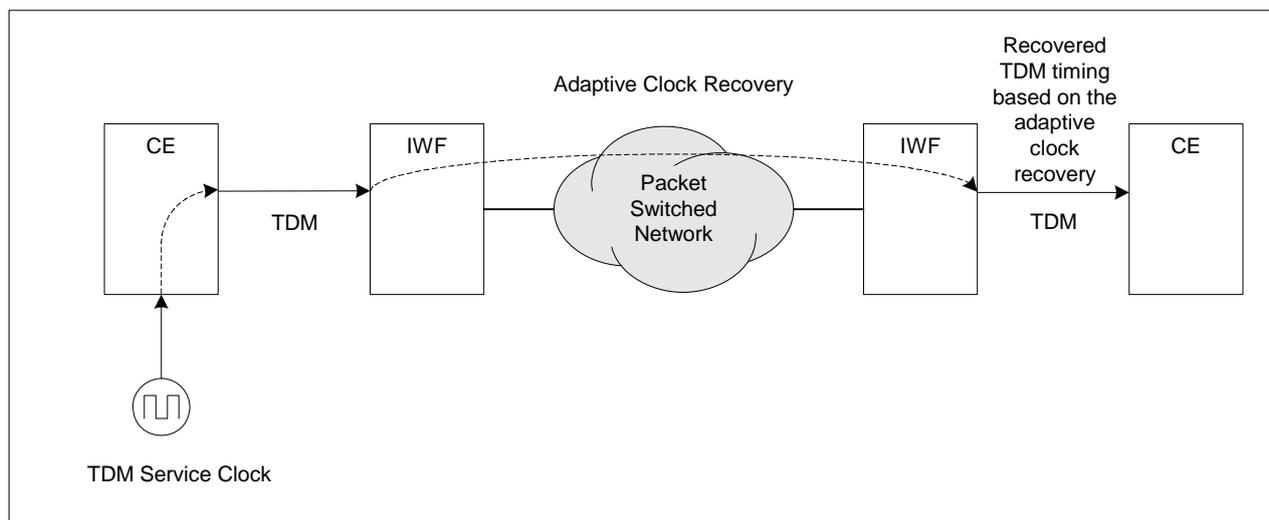


Figure 6 – Example of adaptive method

8.4 Reference clock available at the TDM end systems

When a reference clock is available at each TDM end system, this is a trivial case, since both the end systems have direct access to a timing reference, and will retime the signal leaving the IWF. Therefore, there is no need to recover the timing.

The use of loop timing in the IWF on the TDM interface is an example of the implementation of this method (see Figure 7). An example, of when this scenario might apply, is when two PSTN domains are connected via a packet network. In this case, both the transmitter and receiver are digital switches where there is a need to control slips.

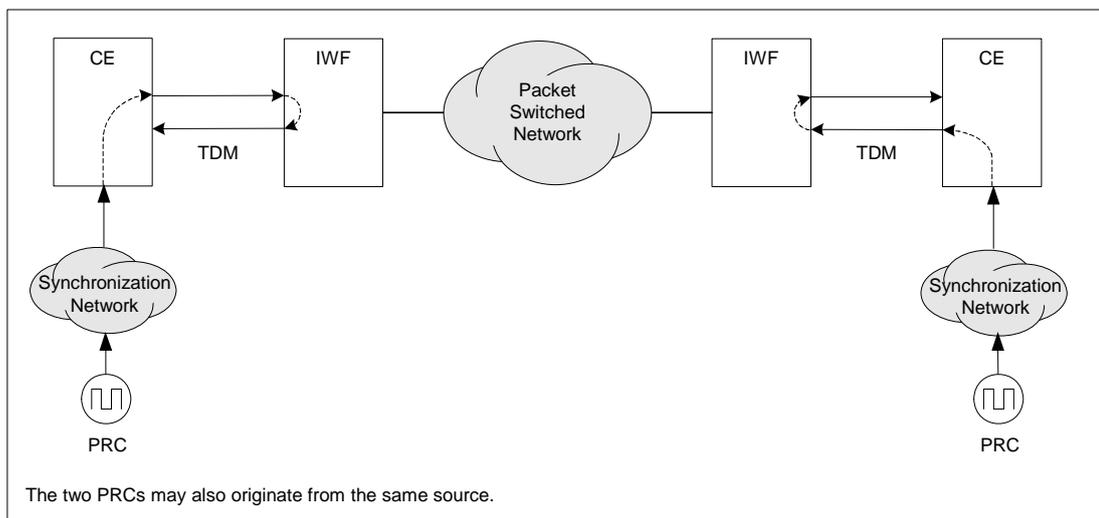


Figure 7 – Example of PRC reference timing signal available at the TDM end systems

9 Network limits

9.1 CES network limits

The network limits on the TDM links at the output of the CES IWF (output of the Reference selector in the CES IWF in Figure B.4) are defined in this clause.

The jitter and wander network limits currently specified in the relevant ITU-T Recommendations (i.e., [ITU-T G.823] and [ITU-T G.824]) have to be fulfilled in all the scenarios that are relevant for this Recommendation.

This clause describes three different deployment scenarios for a CES segment or island. The jitter and wander limits for TDM traffic interfaces (excluding STM-N signals) carried over the CES segment in each of these scenarios are defined in this clause.

The network limits, applicable to synchronization interfaces (as specified in [ITU-T G.823] and in clause 6 of [ITU-T G.824]) and to STM-N signals carried over packet networks, are for further study.

It should be noted that in some cases, signals with quality, according to clause 5 of [ITU-T G.823] and clause 5 of [ITU-T G.824] (traffic interfaces), when traceable to a PRC, can be used as reference timing signals towards an end equipment able to tolerate these signals and to correctly operate (model for deployment case 2 is an example of this scenario).

NOTE – The network limits provided by this clause shall be valid under normal conditions (e.g., when failure conditions or maintenance actions are not present). It is out of the scope of this Recommendation to specify the proportion of time during which these limits apply.

9.1.1 Network model underlying the network limits

For the transport of PDH signals, the models in Figure A.1 of [ITU-T G.823] and in Figure A.1 of [ITU-T G.824] are the starting point to consider the insertion of a CES segment. The wander budget allocation for the CES segment must be only a portion of the entire wander budget as specified in [ITU-T G.823] or [ITU-T G.824], since the total wander budget has to be shared with the rest of the network.

Depending on where the CES segment is located, different wander requirements may apply. Several models of CES deployment have been identified; the models are defined in clauses 9.1.1.1, 9.1.1.2 and 9.1.1.3.

NOTE 1 – The figures in this clause do not show the details on how the timing is recovered by the IWF or how the timing is distributed in the packet network. For further details, refer to clauses 7 and 8.

NOTE 2 – Only one CES island is presented in these models, since it aims at allocating wander budget only for the CES technology segment. There might be several CES systems as long as their accumulated wander generation is within the budget allocated for CES.

The wander accumulation through multiple islands is for further study.

9.1.1.1 Deployment Case 1

When the CES segment is located at an island between the two switches of the G.823 reference model, the wander budget is calculated based on the model in Figure 8. The model is based on Figure A.1 of [ITU-T G.823], and Figure A.1 of [ITU-T G.824] where one of the SDH islands is replaced by the CES network.

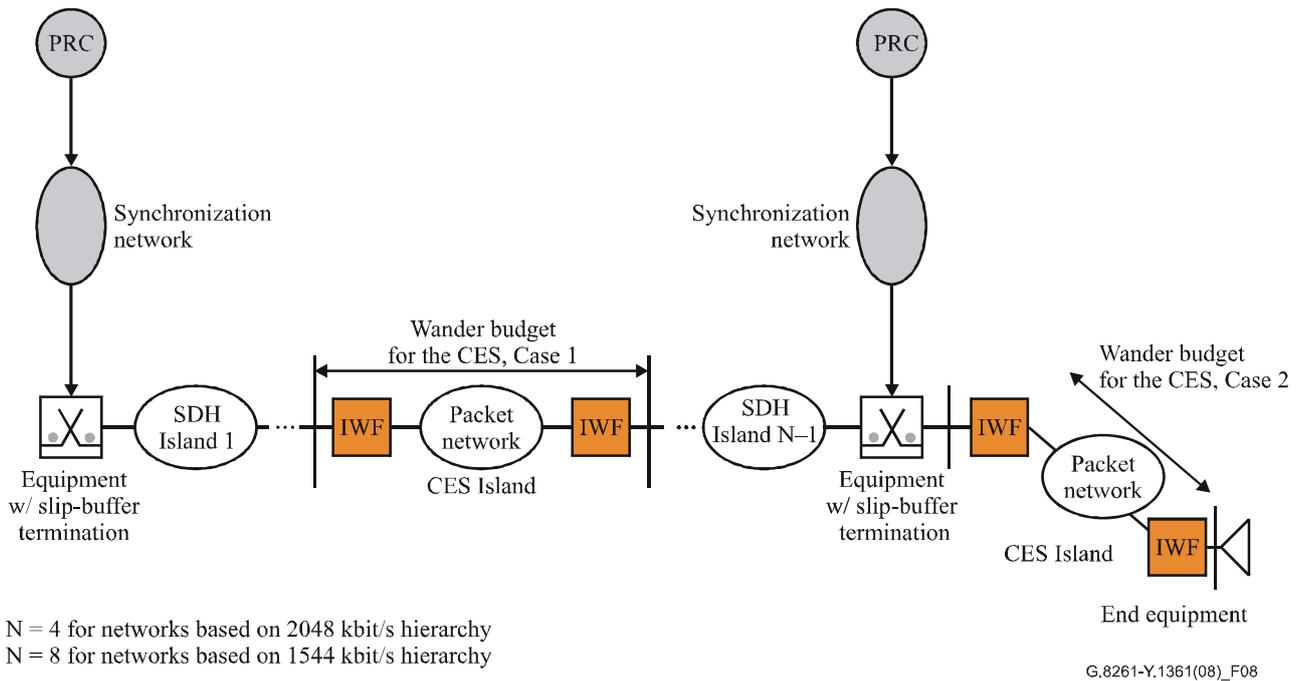


Figure 8 – Network models for traffic and clock wander accumulation, Deployment Case 1 and Case 2

The wander budget for 2048 kbit/s signal is defined in Table 1. The resultant overall specification is illustrated in Figure 9.

Table 1 – Deployment Case 1: 2048 kbit/s interface wander budget

Observation Interval τ (s)	MRTIE requirement (μ s)
$0.05 < \tau \leq 0.2$	10.75τ
$0.2 < \tau \leq 32$	$9 * 0.24 = 2.15$
$32 < \tau \leq 64$	0.067τ
$64 < \tau \leq 1000$	$18 * 0.24 = 4.3$

Note that for the asynchronous configuration, the maximum observation interval to be considered is 80 seconds.
The specification between 80 s and 1000 s for the asynchronous interfaces is for further study.

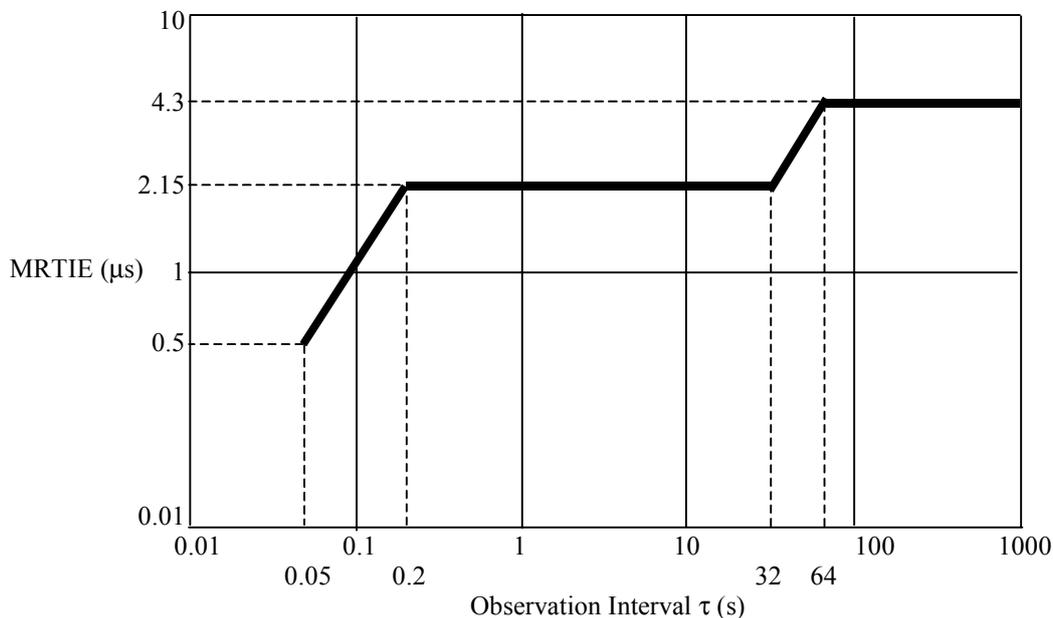


Figure 9 – Deployment Case 1: 2048 kbit/s interface wander budget

The 2048 kbit/s jitter network limits shall comply with clause 5.1 of [ITU-T G.823].

The wander budget for 1544 kbit/s signal is defined in Table 2. The resultant overall specification is illustrated in Figure 10.

Table 2 – Deployment Case 1: wander budget for 1544 kbit/s interface

Observation interval (τ) in seconds	MTIE in μ s
$\tau \leq 0.1$	No Requirement (see Note)
$0.1 < \tau \leq 0.47$	4.5τ
$0.47 < \tau \leq 900$	2.1
$900 < \tau \leq 1930$	$2.33 * 10e-3 \tau$
$1930 < \tau \leq 86\ 400$	4.5

NOTE – This region is covered by jitter requirements.

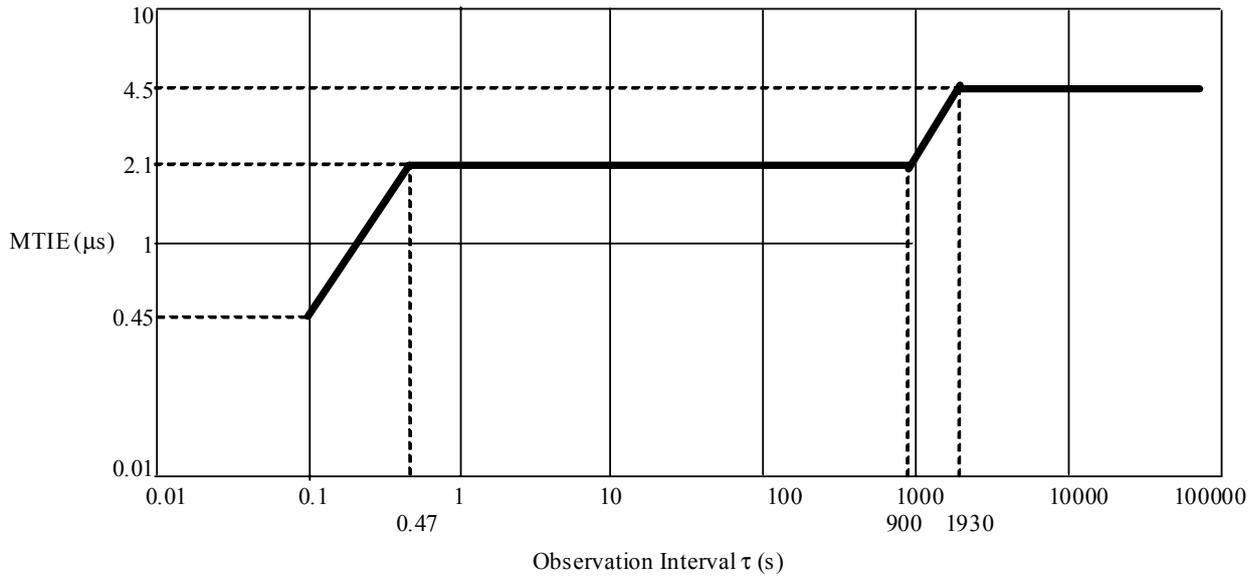


Figure 10 – Deployment Case 1: wander budget for 1544 kbit/s interface

The 1544 kbit/s jitter network limits shall comply with clause 5.1 of [ITU-T G.824].

NOTE – The network limits for the other PDH signals (i.e., 34 368 kbit/s, 44 736 kbit/s and 139 264 kbit/s signals) carried over the CES segments are for further study.

9.1.1.2 Deployment Case 2

Application A

When the CES segment is located outside the network elements containing the slip buffers (see Figure 8), the retiming effect of the switch has to be considered. At the output of this equipment, the timing of the traffic signal will meet the network limit for a synchronization signal which is tighter than for a traffic signal.

The jitter and wander budget of the CES segment in this case is the difference between the 2048 kbit/s network limit (see Figure 1 of [ITU-T G.823]) and the 2048 kbit/s synchronization interface network limit (see Figure 10 of [ITU-T G.823]). The limit is provided in Table 3. The resultant overall specification is illustrated in Figure 11.

Table 3 – Case 2A: 2048 kbit/s interface wander budget

Observation Interval τ (s)	MRTIE requirement (μ s)
$0.05 < \tau \leq 0.2$	40τ
$0.2 < \tau \leq 32$	8
$32 < \tau \leq 64$	0.25τ
$64 < \tau \leq 1000$ (Note)	16
Note that for the asynchronous configuration, the maximum observation interval to be considered is 80 seconds. The specification between 80 s and 1000 s for the asynchronous interfaces is for further study.	

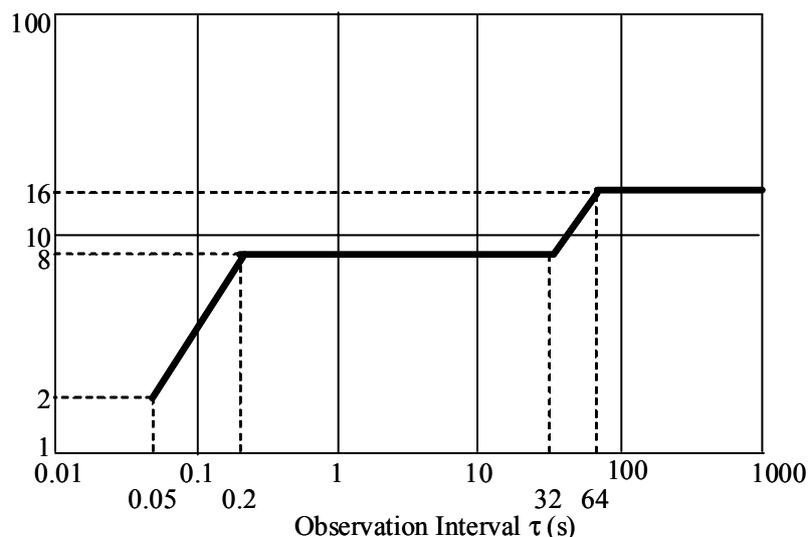


Figure 11 – Case 2A: 2048 kbit/s interface wander budget

In the case of 1544 kbit/s interfaces, Case 1 requirements also apply to Case 2 applications.

NOTE 1 – The network limits for the other PDH signals (i.e., 34 368 kbit/s, 44 736 kbit/s and 139 264 kbit/s signals) carried over the CES segments are for further study.

Application B

In this case, the application recovers timing through the TDM signal; therefore, there is no differential jitter and wander between the clock and the data other than within the bandwidth of the clock recovery since the data and clock are extracted from the same signal. The wander budget of the CES segment is only limited by the timing quality requested by the application (e.g., base station requirements) and not necessarily by [ITU-T G.823].

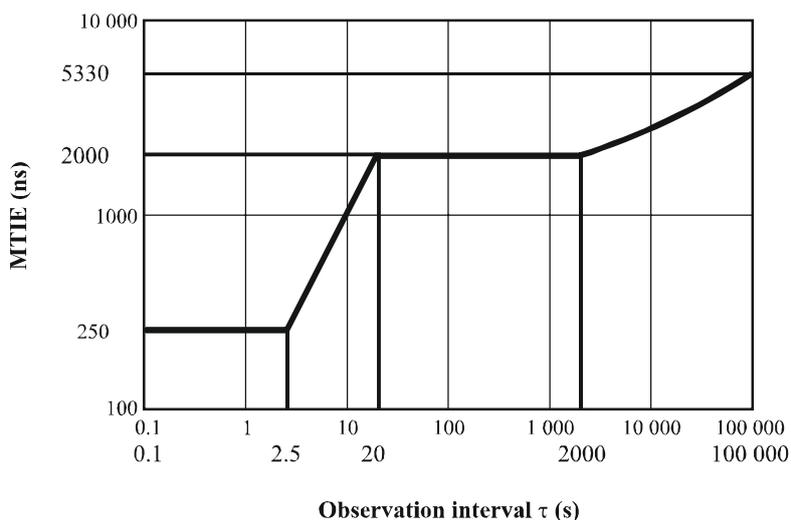
NOTE 2 – This application is valid only for applications with a single signal; if instead two signals are received, then the jitter and wander for one of those signals may differ from that of the clock extracted from the other signal.

9.1.1.3 Deployment Case 3

When retiming is implemented at the output of the SDH islands as shown in Figure 12, the amplitude of noise on the PDH output is that of a synchronization interface. This allows an increase in the wander budget up to that of case 2 Application A in some configurations. It should be noted that the service clock is not preserved end-to-end in this case.

**Table 4 – Network limit for wander at
EEC-Option 1 interfaces expressed in MTIE**

Observation interval τ (s)	MTIE requirement (ns)
$0.1 < \tau \leq 2.5$	250
$2.5 < \tau \leq 20$	100τ
$20 < \tau \leq 2000$	2000
$\tau > 2000$	$433 \tau^{0.2} + 0.01 \tau$



G.8261-Y.1361(08)_F13

**Figure 13 – Network limit for wander (MTIE)
at EEC-Option 1 interfaces**

The network limit for wander at the output interface of an EEC-Option 1, expressed in TDEV, is given in Table 5. The resultant overall specification is illustrated in Figure 14.

**Table 5 – Network limit for wander at
EEC-Option 1 interfaces expressed in TDEV**

Observation interval τ (s)	TDEV requirement (ns)
$0.1 < \tau \leq 17.14$	12
$17.14 < \tau \leq 100$	0.7τ
$100 < \tau \leq 1\ 000\ 000$	$58 + 1.2 \tau^{0.5} + 0.000\ 3 \tau$

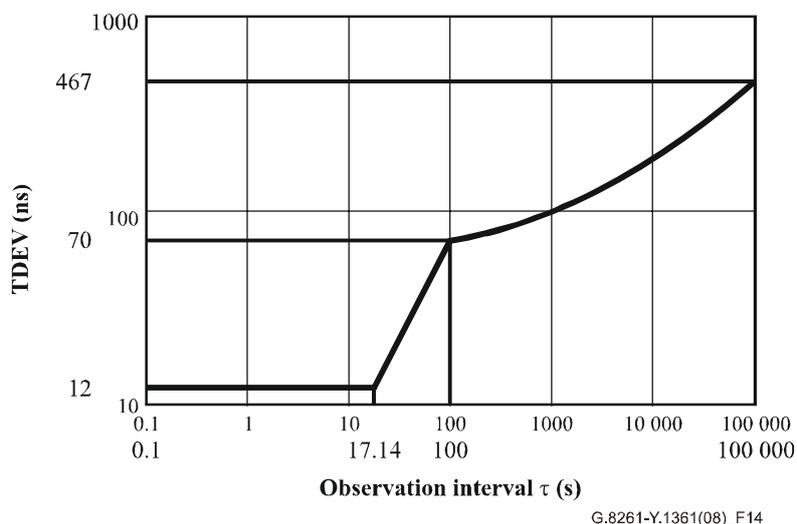


Figure 14 – Network limit for wander (TDEV) at EEC-Option 1 interfaces

9.2.1.2 EEC-Option 2 interface network wander limits

The network limit for wander at the output interface of an EEC-Option 2, expressed in TDEV, is given in Table 6. The resultant overall specification is illustrated in Figure 15.

Table 6 – Network limit for wander (TDEV) at EEC-Option 2

Observation interval, τ (s)	TDEV (ns)
$0.05 < \tau \leq 10$	10
$10 < \tau \leq 1000$	$3.1623 \tau^{0.5}$

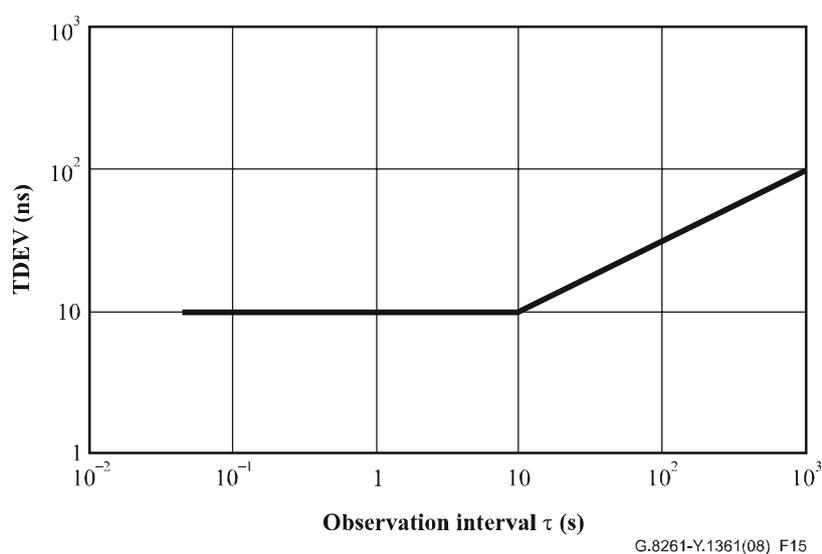


Figure 15 – Network limit for wander (TDEV) at EEC-Option 2

The mask in Figure 15 is taken from Figure 5 of [ITU-T G.824]. This mask is also found in Figure I.1 of [ITU-T G.813] for Option 2 Network wander limits.

9.2.1.3 EEC interface network jitter limits

See Table 7 for EEC interface network jitter limits.

Table 7 – EEC interface network jitter limits

Interface	Reference	
2048 kbit/s	See G.823 clause 6.1: Network limits for output jitter at synchronization interfaces, SEC requirements	(Note 1)
2048 kHz		
1544 kbit/s		
STM-n		
Ethernet (synchronous Ethernet)	For further study	(Note 2)

NOTE 1 – Jitter limits are taken from G.823, G.824 and G.825 in order to allow proper interoperability with SEC-based synchronization networks and combined EEC-SEC functions.

NOTE 2 – In a chain of n ($n \leq 20$) connected EECs, the accumulated network jitter has to be low enough to allow all involved EECs to meet the output jitter specification at their synchronization outputs (e.g., 2048 kHz, 2048 kbit/s, 1544 kbit/s). See Figure 16 showing EECs in a chain; see also Annex D.

Figure 16 shows the reference chain of n ($n \leq 20$) EECs together with their synchronization outputs.

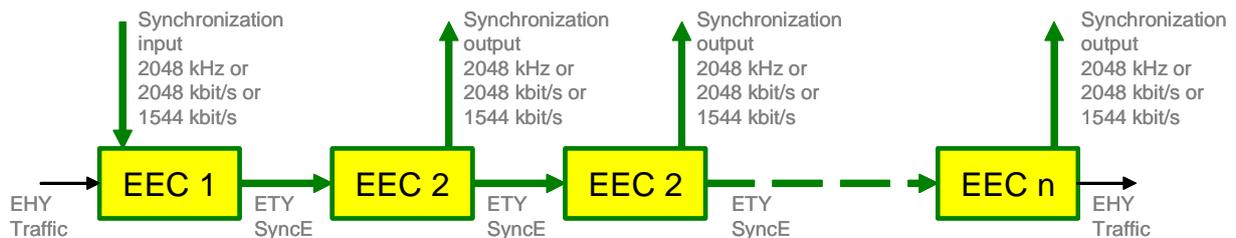


Figure 16 – EEC chain

9.2.2 PEC interface network limits

The network limits at the output of the PEC (see Figure B.5) are defined in this clause.

This clause describes two different deployment scenarios for a PNT segment or island with packet-based equipment clocks (PECs). The jitter and wander limits for the PEC interfaces are defined in this clause for each of these scenarios.

9.2.2.1 Network model underlying the PEC network limits

For the transport of reference timing signals, the models in Figure 8-5 of [ITU-T G.803] and in Figure B.3 of [ITU-T G.823] are the starting point to consider the insertion of a PNT segment.

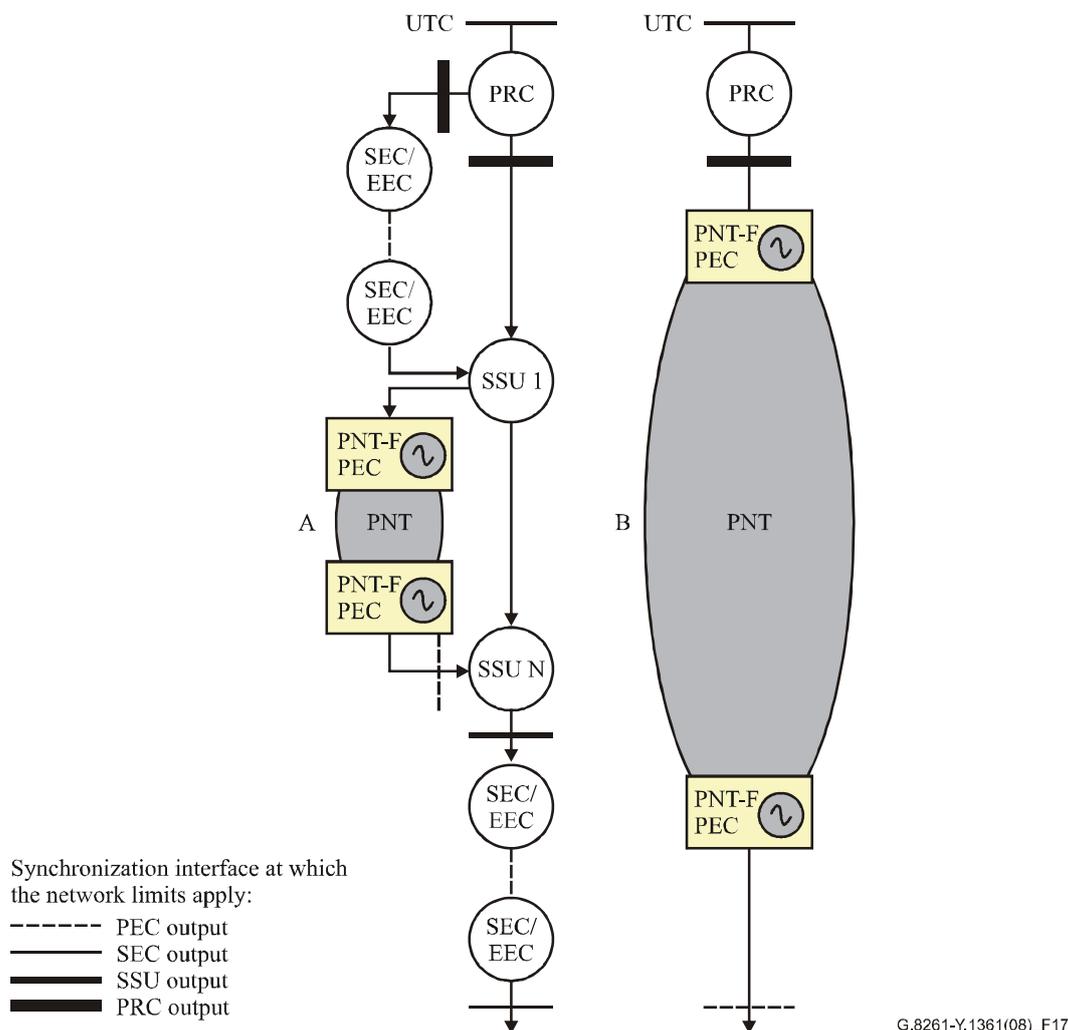
Depending on where the PNT segment is located, different wander requirements may apply. Several models of PNT deployment have been identified; the models are defined in this clause.

NOTE – The figures in this clause do not show the details on how the timing is distributed in the packet network. For further details, refer to clause 7.

PNT Deployment Case 1

The model for the PNT deployment Case 1 is shown in Figure 17. This case relates to a master-slave synchronization network that, instead of using TDM based technologies (e.g., SDH or PDH), is implemented over packet network technologies.

The left figure (model A) shows an example where part of the synchronization network is replaced by a PNT segment, and the right figure (model B) refers to a synchronization network fully implemented over a PNT segment.



NOTE – The PEC is the PNT-F clock that is relevant in this model. The deployment of other clocks in the network (e.g., deployment of part of the PNT segment based on EEC and part on PEC) is for further study.

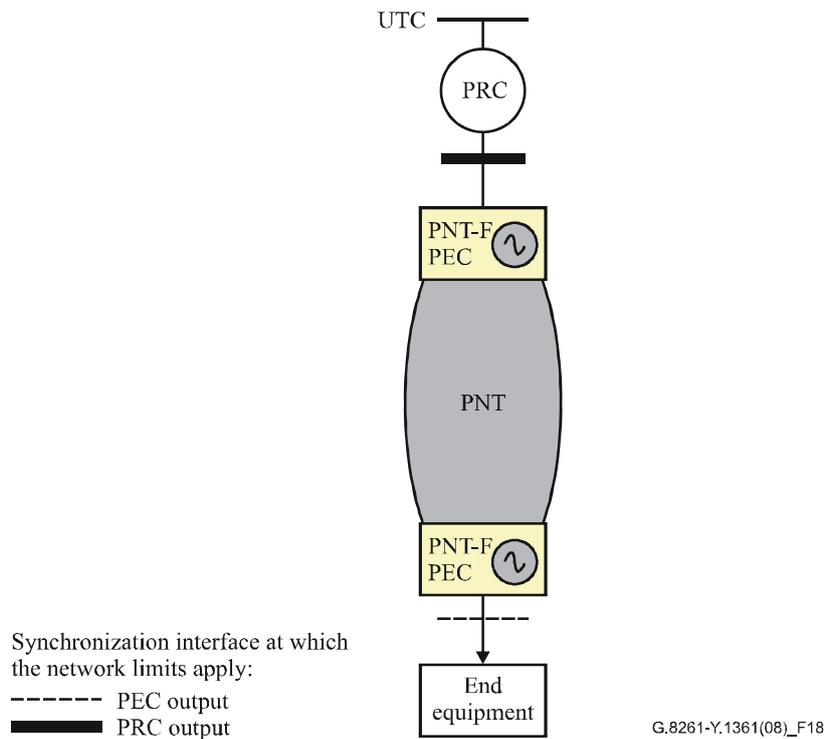
Figure 17 – PNT segment replacing part (model A) or all (model B) of the TDM-based synchronization network

The network limits as defined in clause 9.2.1 (EEC network limits) apply in both model A and B.

PNT Deployment Case 2

The model for the PNT deployment Case 2 is shown in Figure 18.

This case is related to distributing the timing towards end equipment application (e.g., BTS, see Appendix IV).



NOTE – The PEC is the PNT-F clock that is relevant in this model. The deployment of other clocks in the network (e.g., deployment of part of the PNT segment based on EEC and part on PEC) is for further study.

Figure 18 – PNT distributing timing towards end applications

In deployment case 2, the requirements are set by the end equipment. These are expressed in terms of tolerance and in terms of level of accuracy that the application requires for the correct operations.

Therefore, the PNT segment in deployment case 2 is allowed to generate jitter and wander up to the limits tolerated by the end applications (e.g., network limits for traffic interfaces as defined by clause 5 of [ITU-T G.823] or clause 5 of [ITU-T G.824]).

10 Impact of impairments in the packet network on timing distribution and service clock recovery

This clause discusses the different impairments impacting the traffic and its timing information in packet networks. It is understood that the requirements of emulated circuits and recovered clocks, which are specified in clause 9, are to be met under operational conditions.

Fundamentally, network synchronization is required in layer 1 networks to manage buffers. Layer 1 buffers as present in PDH, SDH and OTN networks and its adaptation functions are simple structures, where the nominal ingress and egress rate is controlled within specific bounds as given in the related networking standards of these TDM networks. Mechanisms, such as stuff bytes and pointers, together with system clocks, are the used methods to manage these buffers and accommodate different clocking domains. The network design constrains the buffer size to minimize latency. In layer 1 networks, such as SDH, there is a direct linkage between the network clock and the level of wander or jitter that may be imparted on a client signal.

In the case of packet data networks transport, the data is delivered over the network in blocks (packets, frames), rather than being carried as a continuous stream of constant bit rate. Packets may be statistically multiplexed and are routed via packet switches which subject a packet to delay due to processing, buffering and retransmission at intermediate switches. Within a single switch, multiple

packet streams may have to converge onto a single output buffer. The resulting buffer contention will introduce variable delay. In some cases, packets will be dropped. The clock used to drive the layer 1 transmission links is likely to be asynchronous to the clock used within the switch. Any difference in the rate at which packets are presented for transmission, and the actual transmission rate is accommodated by adding padding between packets or discarding packets.

Since packets may traverse different routes, a stream of packets from ingress to egress may exhibit significant packet delay variation. Additionally, packets may be misordered resulting in additional buffering. Services that utilize the packet network need to consider these impairments. For packet networks, large buffers are required to perform packet level processing, in which case only coarse levels of synchronization are required to support most services.

Unlike a layer 1 network, such as SDH, there is no direct linkage between the network clock and the packet processing buffers. Thus, network timing cannot be used to control packet delay variation in these networks. The need to provide network synchronization to a packet switch is generally required only to meet any synchronization requirements for physical layer interfaces to the switch in accordance to the related TDM interface requirements, as given by the particular networking standards as SDH/PDH.

Timing requirements of services carried at upper layers above the layer 2 network (e.g., IPTV, MPEG-4) are specified to accommodate the variations of existing packet networks. Any service specific timing is encoded at the service layer (e.g., H.264, MPEG-4).

However, there are cases when the physical layer of a packet network is synchronous (e.g., SDH) and may be utilized by the adaptation layer.

In most cases, the information carried across the packet network, i.e., the characteristic information, does not include timing information. This has certain ramifications when services require the transfer of accurate timing. For end-to-end services, the timing characteristics of the server layer need to support the synchronization requirements of the client. In traditional layer 1 mechanisms (PDH, SDH and OTN), the network timing adaptation mechanisms are specifically designed to be compatible with the timing requirements of the client signal. When the server layer is incapable of supporting the timing of the client, alternate means of providing client timing may be required. This would be performed at the adaptation layer to the network. An example is ATM AAL1.

Impairments in the packet network may have a detrimental effect on the recovery of the service a clock using adaptive methods. This clause explores the levels of such impairments that a clock recovery process should be capable of withstanding, while maintaining the clock within the relevant specifications.

The following performance parameters relating to packet network impairments are defined in [ITU-T Y.1540] (for IP networks) and [ITU-T Y.1561] (for MPLS networks). Similar performance measures for Ethernet networks are also defined in [ITU-T Y.1731].

- 1) Packet transfer delay and delay variation
- 2) Packet error ratio
- 3) Packet loss ratio
- 4) Packet severe loss block outcomes

10.1 Packet transfer delay and delay variation

10.1.1 Differential methods

Packet transfer delay and delay variation should not affect clock recovery performance when a network reference clock is available at both ends and differential methods are used.

10.1.2 Adaptive methods

Adaptive recovery of the service clock from a packet stream containing constant bit rate or timestamp data is, in general, achieved by some computing function of the arrival rate or arrival times of the packets at the destination node.

If the delay through the packet network is constant, the frequency of arrival of packets at the destination node is not affected by the network. There may be a phase lag in the recovered clock due to the delay through the network, but there should be no frequency or phase wander.

If the delay varies, it may be perceived by a clock recovery process as a change in phase or frequency of the original service clock. Therefore during the design of a clock recovery process, the causes of delay variation must be considered carefully.

There are several causes of delay variation on a packet network. These may include:

- Random delay variation (e.g., queuing delays)
- Low frequency delay variation (e.g., day/night patterns)
- Systematic delay variation (e.g., store-and-forward mechanisms in the underlying transport layer)
- Routing changes
- Congestion effects

10.1.2.1 Random delay variation

Random variation in delay results from the behaviour of switches or routers in the packet network. The primary source of this is output queuing delay, caused when a packet arrives at a switch or router when the exit port is blocked by other traffic, and the packet has to wait in a queue. Other factors caused by the internal operation of the switch or router may also delay the packet, as described in Appendix I.

It is not possible to predict with any certainty the delay of any packet through a switch or router, although it is likely that the delay will increase with load on the device. Therefore, there will be some correlation of delay between successive packets with the traffic load in the network.

10.1.2.2 Low frequency delay variation

As described above, the delay through a packet network, although unpredictable, is generally correlated with the load on the network at the time period in question. Load is a dynamic quantity, and may contain extremely low frequency components. For example, if a network is more heavily loaded during the day than at night, this gives a component to the load variation with a 24-hour period.

This extremely low frequency variation may lead to phase wander in a clock recovered from a packet stream with the same period. Since many of the relevant clock specifications limit the permissible phase wander over periods of 24 hours or more (e.g., [ITU-T G.824]), this has to be compensated for during the design of a clock recovery process.

10.1.2.3 Systematic delay variation

Certain types of underlying transport networks can cause systematic variation in packet delay over time. For example, some types of transport use a "transmission window" or "timeslot", storing packets for transmission until the window opens. Examples of these include PON, xDSL and WiMAX.

The effect of the transmission window is to impose a systematic "sawtooth" delay profile onto a packet stream (see Figure 19). For regular rate packet streams, e.g., those containing constant bit rate data, the period of the transmission window may beat against the packet rate, causing a slow variation in delay over time. These effects are very similar to the waiting time jitter in TDM

networks. In TDM networks, it is possible to control the waiting time jitter while this is not the case in packet networks.

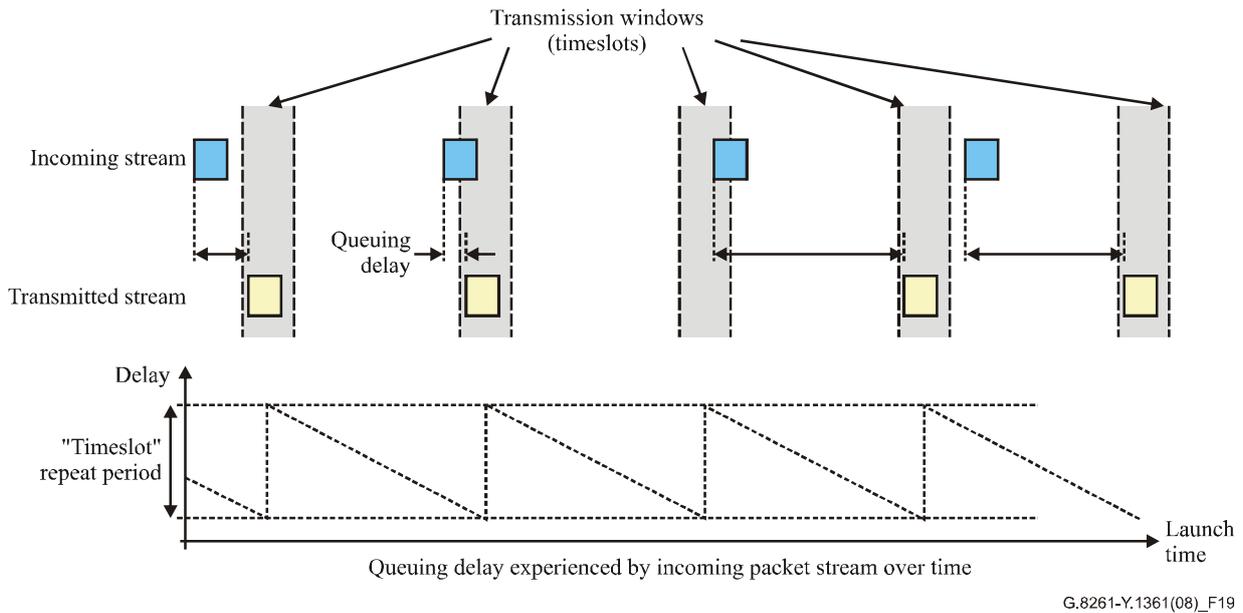


Figure 19 – Systematic delay variation caused by network with timeslots

Another type of systematic delay variation that regular rate packet streams may be subject to is beating against other regular packet streams. Figure 20 shows what happens when two packet streams of almost the same frequency are combined onto a single packet link by a switch or router.

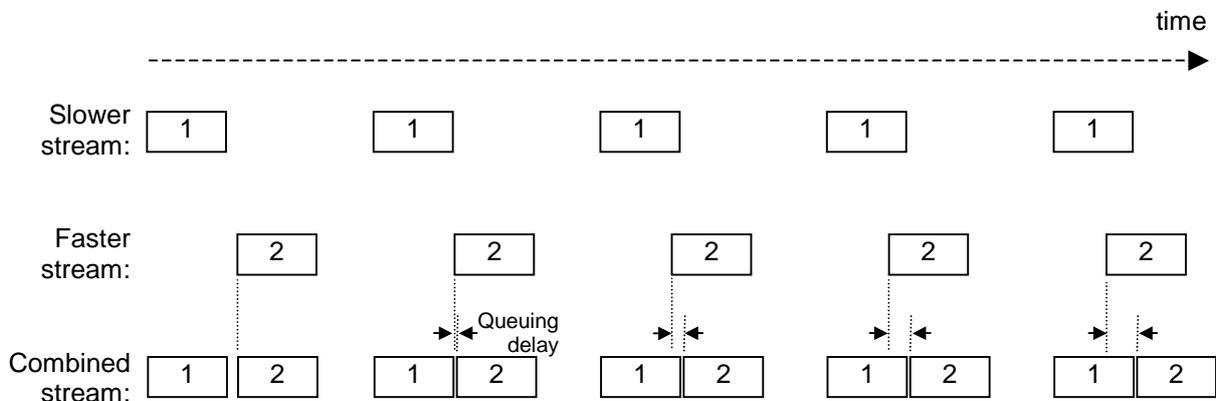


Figure 20 – Beating between regular rate packet streams

Stream 1 is the slower stream, and for a time the packets in stream 1 arrive at the switch or router ahead of those in stream 2. However, the packets in stream 2 start to catch up. Since only one packet at a time can be output onto the packet link, the packets in stream 2 start to experience queuing delay (see Figure 21). This delay builds up to the point where it is equal to the transmission time of the packet on the link.

Eventually, the packets in stream 2 start to arrive at the switch or router ahead of those in stream 1, and the queuing delay is removed. At this point, it is stream 1 which now sees a queuing delay. This steadily declines until the packets in stream 1 arrive at the switch after the packets in stream 2 have completed transmission.

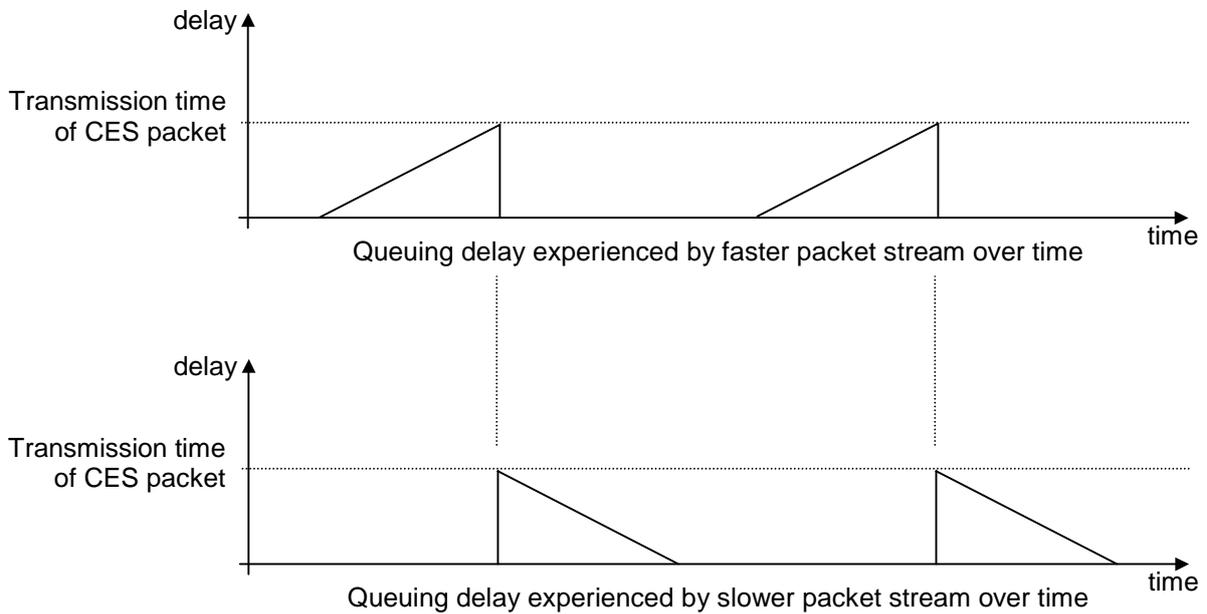


Figure 21 – Delay profile experienced by beating packet streams

The duration of time that the packet streams experience queuing delay (i.e., the width of the triangles in Figure 21) is inversely proportional to the difference in rate between the two packet streams. Where the packet rates are very close, the duration may be extremely long. This long-term variation in the delay may cause a slow phase wander in any clock recovered from one of the packet streams.

Where multiple asynchronous regular rate streams share the same packet link, the effect may be additive. In the worst case, packets from all streams may line up exactly yielding the maximum queuing delay, although the frequency of this combined beat will reduce with the number of streams.

10.1.2.4 Routing changes

The route taken by a stream of packets through a packet network may change at certain instants in time. This may be due to network errors (e.g., routing around a failed or congested link), protection switching to use an alternative route, or network reconfiguration.

The net effect of this is a step change in delay through the network. If uncompensated, this may be seen in the recovered clock as a phase change. Such changes must be detected and accounted for in the clock recovery process. In general, large changes in delay are relatively easy to detect and compensate for, but small changes may be masked by the general delay variation, or local oscillator drift at the clock recovery node.

10.1.2.5 Congestion effects

Congestion is the temporary increase in traffic load in all or part of a network. It may lead to all or part of the network becoming "overloaded", and packets becoming severely delayed or dropped. The duration of congestion events is variable, and may last for several seconds or minutes. If the network experiences frequent severe congestion events lasting longer than 5 minutes, it is an indication that the network is probably not suitable for running circuit emulation.

10.1.2.6 Topology-dependent blocking mechanisms in packet networks

The topology of the network and, in particular, the interaction with other flows in the network can affect the delay of a packet flow. This is due to the fact that different size packets traverse the network at different rates. Large packets take longer to traverse a network, because they have to be fully read into a network element before they can be processed and forwarded to the next hop. CES and timing packets are typically short and move quickly through a network.

In a network with a shared route topology (e.g., as shown in Figure 22), this can cause additional delay, even at quite low traffic volumes.

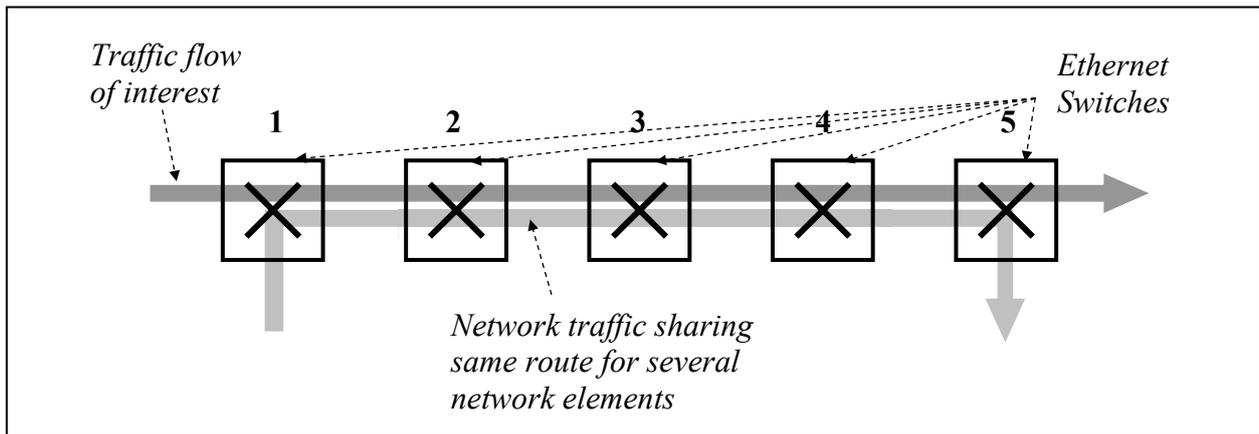
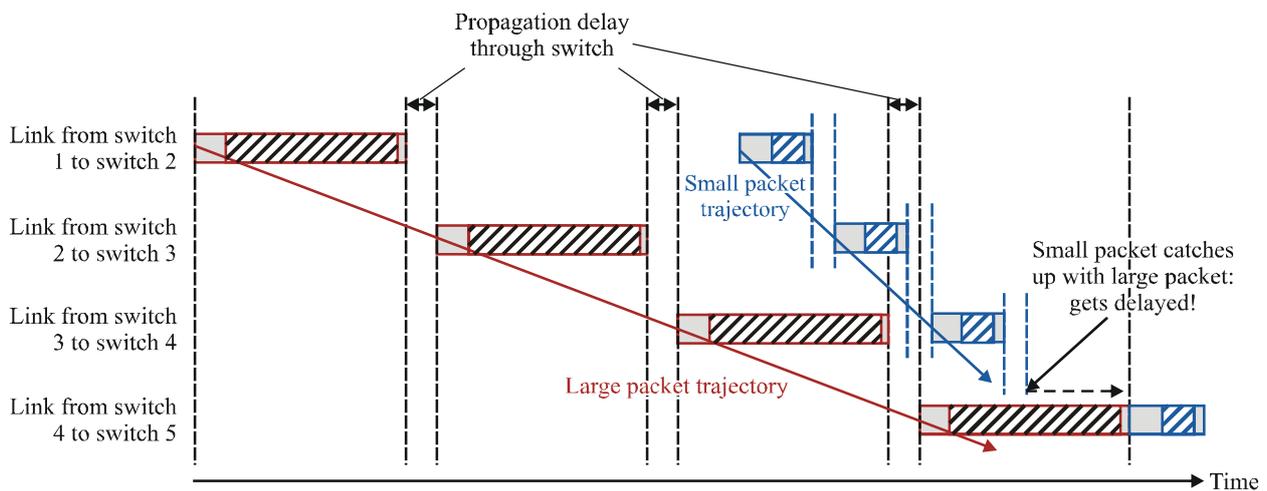


Figure 22 – Shared route topology

Where a large packet flow shares the same path for two or more consecutive links, it can begin to obstruct the transmission of small packets. This is where small packets "catch up" with large packets, because they propagate through the network quicker. Figure 23 shows how the effect works:



G.8261-Y.1361(08)_F23

Figure 23 – Large packet blocking along shared route

The effect is such that if load of big packets is sufficiently high, the small packets are likely to be delayed at some point in their transit through the network (note this changes with the relative size of the small to large packets). It causes the minimum possible delay through the network to increase proportionally above this load.

10.2 Impacts from packet impairments

10.2.1 Packet error and packet loss

Impairments within packet networks have an impact on three distinct elements within the delivery path, the IWF clock recovery process (note this may not be available to monitor), the service clock recovery and the TDM service itself. Packet loss and packet misordering limits and their impact on the service and clock recovery processes are for further study.

Additional text discussing the relevant issues is given in the following subclauses.

Packet loss and packet misordering do not significantly affect the IWF clock recovery performance for any of the methods presented in this Recommendation. Specifically, at levels at which the TDM transport service remains useable, packet loss (both uniform and bursty) and packet misordering have negligible effect on IWF clock recovery performance.

10.2.1.1 Impact on TDM service

TDM circuits carried over packet networks may be extremely vulnerable to bit errors caused by packet loss. One of the reasons for this is that bit errors are magnified by the packet transport – a single bit error in the packet leads to the whole packet being discarded, yielding a burst of consecutive bit errors in the recovered TDM stream. Hence, even moderate levels of packet loss (from the viewpoint of conventional packet network) may cause unavailability of a TDM circuit.

NOTE – The vulnerability of the TDM circuits would mainly depend upon the specific IWF characteristics. Some IWFs might employ different packet-loss concealment techniques to protect the application from packet loss.

10.2.1.2 Impact on IWF clock recovery process

The IWF clock recovery combines the packet to clock recovery algorithm, the embedded clock and the timing recovery method used (i.e., adaptive or differential). Performance of the IWF clock recovery process is a combination of packet network stress, the algorithm used to overcome network stress, the clock embedded within the IWF and the timing recovery method used.

NOTE – The packet loss and misordering limits for the preservation of IWF clock recovery and service clock recovery shall be specified to cover all possible packet loss scenarios; these limits are for further study.

10.2.1.3 Impact on service clock recovery

With respect to the service clock recovery process, it is required that the clock recovery has to withstand much higher packet losses than the TDM circuit itself, such that the service clock remains within specification beyond the point where the data is declared unavailable. The IWF clock recovery will directly influence the service clock recovery performance.

10.2.2 Packet severe loss block outcomes

[ITU-T Y.1540] and [ITU-T Y.1561] define a severe loss block outcome as occurring when for a block of packets observed at an ingress interface during time interval T, the ratio of lost packets to total packets exceeds a threshold. Similar effects are expected in Ethernet networks.

During these impairments, the timing recovery mechanism has to handle the total loss of packets as discussed in clause 10.2.1. This subject is for further study.

11 Impact of the reference clock impairment on the service clock recovery

11.1 Impairments for the network synchronous operation methods

The clocks involved in the transport of TDM signals through a packet network are shown in Figure 24.

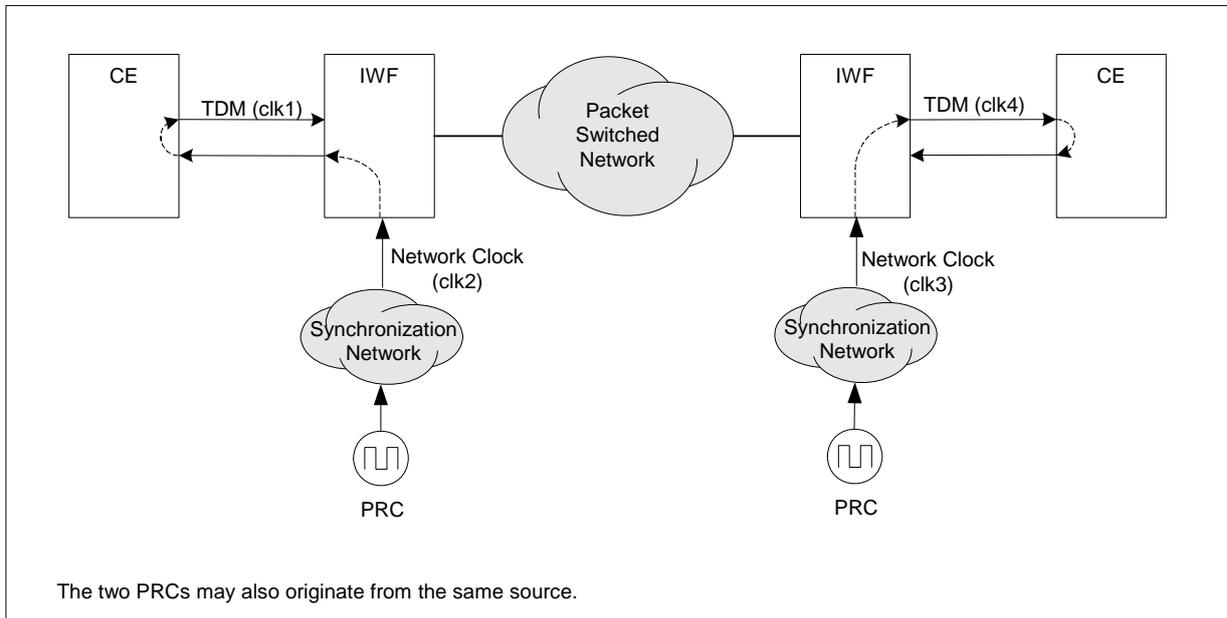


Figure 24 – Clocks involved in the transport of TDM signals through a packet network for network synchronous operation

The clocks are:

- The clock that generates the TDM signal (clk1 in the figure)
- The network reference clock used for de-packetization in the left hand IWF (clk2 in the figure)
- The network reference clock used for de-packetization in the right hand IWF (clk3 in the figure)
- The clock that generates the TDM signal after the packet network (clk4 in the figure)

clk1 has to be traceable to a PRC, this can be either obtained by loop timing as shown in Figure 24, or by other means. If not, the use of a network clock reference in the de-packetizer (i.e., clk3 in the figure) will raise severe problems.

In order to have correct timing in the output TDM signal, the clocks generating (i.e., clk1) and retiming (i.e., clk4) the TDM signals must have the same long-term frequency (or within the PRC limits); otherwise, an unacceptable rate of slips will be generated (the short-term noise shall be kept within the applicable limits).

In normal operation, the network reference clock at the TDM source (clk1) and the network reference clock at the de-packetizer are both locked to a reference timing signal that is traceable to a PRC. However, during failure conditions in the synchronization network, these clocks may be locked to a reference timing signal that is traceable to a clock operating in holdover mode. During failure conditions, these clocks shall provide a suitable holdover that is based on the [ITU-T G.822] slip performance objectives.

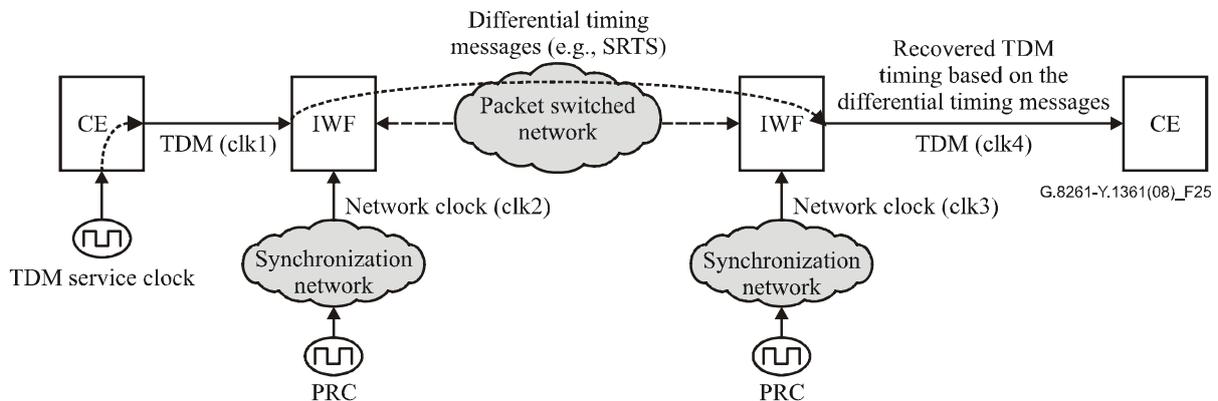
The clock providing this holdover function during failures in the synchronization network may be either integrated in the equipment itself or available in the site (for instance, integrated in a transmission network element or in a SASE). It is the responsibility of the network planner to provide the most suitable solution.

To summarize, the network synchronous mode of operation requires either the introduction of precise clocks in the sink IWF or a system that allows to switch to another suitable clock in case of loss of synchronization from the network clock (PRC).

In order to detect the periods of loss of synchronization, some kind of supervision of the traceability is needed (e.g., SSM).

11.2 Impairments for the differential method

The clocks involved in the transport of TDM signals through a packet network are shown in Figure 25.



The two PRCs may also originate from the same source.

Figure 25 – Clocks involved in the transport of TDM signals through a packet network for differential method

These are:

- The clock that generates the TDM signal, PDH or SDH (clk1 in the figure);
This clock may be plesiochronous although it is considered that most signals are now synchronous.
- The network clock that is used to generate the differential timing messages (clk2 in the figure).
- The network clock (clk3 in the figure) that is used to regenerate the TDM clock (clk4 in the figure) based on the differential timing messages.

Any phase noise on these clocks will cause phase noise on the timing of the output TDM signal.

In order to have correct timing in the output TDM signal, the clocks generating (i.e., clk1) and retiming (i.e., clk4) the TDM signals must have the same long-term frequency (or within the PRC limits); otherwise, an unacceptable rate of slips will be generated (the short-term noise shall be kept within the applicable limits).

In normal operation, the network clocks generating the differential timing messages and regenerating the TDM clock (clk2 and clk3 in the figure) are locked to a reference timing signal that is traceable to a PRC. However, during failure conditions in the synchronization network, these clocks may be locked to a reference timing signal that is traceable to a clock operating in holdover mode. During failure conditions, these clocks shall provide a suitable holdover that is based on the [ITU-T G.822] slip performance objectives.

The clock providing this holdover function during failures in the synchronization network may be either integrated in the IWF itself or available in the site (for instance, integrated in a transmission network element or in a SASE). It is the responsibility of the network planner to provide the most suitable solution.

In order to detect the periods of loss of synchronization, some kind of supervision of the traceability is needed (e.g., SSM).

12 Results and consequences of the different synchronization methods over packet network reference models

The recommendations on the methodology to distribute synchronization references (PNT domain) and to recover the timing of a TDM service (CES domain) differ according to the network scenarios and to the synchronization requirements that are relevant for the specific application.

12.1 CES domain recommendations

The following scenarios have been identified within the scope of this Recommendation (with reference to the network models in clause 9.1).

12.1.1 Recommendation for the timing recovery of a TDM service (Deployment Case 1)

The network limits for PDH signals in this case are defined in clause 9.1 for Deployment Case 1.

The timing recovery of PDH signals carried over packet network can be performed by means of:

- Network synchronous operation when a signal traceable to PRC is available at the IWFs, and it is not required to preserve the service clock.
- Differential methods when a PRC traceable reference is available at the IWF. With this method, it is possible to preserve the service clock.
- Adaptive methods when the delay variation in the network can be controlled. With this method, it is possible to preserve the service clock.

NOTE – In these scenarios, the network limits are quite stringent. However, it is assumed that when the network can be modelled according to Model A (at least Scenario 2 and Scenario 3, see Appendix V), the adaptive methods should allow the compliance with the network limits as defined in clause 9.1.

It is for further study if the adaptive method can be used in a network that can be modelled according to Model B (see Appendix V).

The transport of SDH signals in this scenario is for further study. It should be noted that the clock recovery for SDH signals shall fulfil the quality level, as per synchronization interfaces according to [ITU-T G.823] for networks based on the 2048 kbit/s hierarchy, and to [ITU-T G.824] for networks based on the 1544 kbit/s hierarchy. The use of the methods as described in clause 7.1 to support a network synchronous operation can guarantee that these requirements are fulfilled.

12.1.2 Recommendation for the timing recovery of a TDM service (Deployment Case 3)

The network limits for PDH signals in this case are defined in clause 9.1 for Deployment Case 3.

The timing recovery of PDH signals carried over packet network can be performed by means of:

- Network synchronous operation when a signal traceable to PRC is available at the IWFs, and it is not required to preserve the service clock.
- Differential methods when a PRC traceable reference is available at the IWF. With this method, it is possible to preserve the service clock.
- Adaptive methods when the delay variation in the network can be controlled. With this method, it is possible to preserve the service clock.

NOTE – In these scenarios, the network limits are less stringent than in the clause 12.1.1 scenarios. It is assumed that, when the network can be modelled according to Model A, the adaptive methods should allow the compliance with the network limits as defined in clause 9.1.

It is for further study if the adaptive method can be used in a network that can be modelled according to Model B.

The transport of SDH signals in this scenario is for further study. It should be noted that the clock recovery for SDH signals shall fulfil the quality level as per synchronization interfaces according to [ITU-T G.823] for networks based on the 2048 kbit/s hierarchy, and to [ITU-T G.824] for networks based on the 1544 kbit/s hierarchy. The use of the methods as described in clause 7.1 to support a network synchronous operation can guarantee that these requirements are fulfilled.

12.1.3 Recommendation for the timing recovery of a TDM service (Deployment Case 2 Application A)

The network limits for PDH signals in this case are defined in clause 9.1 for Deployment Case 2 Application A.

The timing recovery of PDH signals carried over packet network in this case can be performed by means of:

- Network synchronous operation when a signal traceable to PRC is available at the IWFs, and it is not required to preserve the service clock.
- Differential methods when a PRC traceable reference is available at the IWF. With this method, it is possible to preserve the service clock.
- Adaptive methods when the delay variation in the network can be controlled. With this method, it is possible to preserve the service clock.

NOTE – In these scenarios, the network limits are less stringent than in the clause 12.1.1 scenarios. It is assumed that, when the network can be modelled according to Model A, the adaptive methods should allow the compliance with the network limits as defined in clause 9.1.

It is for further study if the adaptive method can be used in a network that can be modelled according to Model B.

The transport of SDH signals in this scenario is for further study. It should be noted that the clock recovery for SDH signals shall fulfil the quality level as per synchronization interfaces according to [ITU-T G.823] for networks based on the 2048 kbit/s hierarchy, and to [ITU-T G.824] for networks based on the 1544 kbit/s hierarchy. The use of the methods as described in clause 7.1 to support a network synchronous operation can guarantee that these requirements are fulfilled.

12.1.4 Recommendation for the timing recovery of a TDM service (Deployment Case 2 Application B)

The network limits for PDH signals in this case are defined in clause 9.1 for Deployment Case 2 Application B.

The timing recovery of PDH signals carried over packet network in this case can be performed by means of:

- Network synchronous operation when a signal traceable to PRC is available at the IWFs, and it is not required to preserve the service clock.
- Differential methods when a PRC traceable reference is available at the IWF. With this method, it is possible to preserve the service clock.
- Adaptive methods when the delay variation in the network can be controlled. With this method, it is possible to preserve the service clock.

NOTE – In these scenarios, the network limits are dependent on the characteristics of the end equipment that normally is able to tolerate the [ITU-T G.823] and [ITU-T G.824] traffic interface limits. It is assumed that, when the network can be modelled according to Model A, adaptive methods should allow compliance to [ITU-T G.823] or [ITU-T G.824] as appropriate.

It is for further study if the adaptive method can be used in a network that can be modelled according to Model B.

The transport of SDH signals in this scenario is for further study. It should be noted that the clock recovery for SDH signals shall fulfil the quality level as per synchronization interfaces according to [ITU-T G.823] for networks based on the 2048 kbit/s hierarchy, and to [ITU-T G.824] for networks based on the 1544 kbit/s hierarchy. The use of the methods as described in clause 7.1 to support a network synchronous operation can guarantee that these requirements are fulfilled.

12.2 PNT domain recommendations

The following scenarios have been identified within the scope of this Recommendation (with reference to the network models in clause 9.2).

12.2.1 Recommendations for reference timing signals distribution over synchronous Ethernet

The network limits for the reference timing signals in this case are defined in clause 9.2.

The deployment of synchronization networks over the physical layer using a layer 1 technique, such as synchronous Ethernet described in clause 7.1.1 provides a method of transporting synchronization which is not subject to any packet based stress (i.e., no impact from packet delay variation).

The network limits as defined in clause 9.2 are met provided that the design of the synchronization network follow the same design rules already defined for synchronization networks based on SDH, i.e., compliance to [ITU-T G.803] synchronization network reference chain, see Figures 8-1 and 8-2 of [ITU-T G.803] where the EEC [ITU-T G.8262] shall replace the SEC. In particular, based on this model, the longest synchronization network reference chain should not exceed 10 SSUs and 60 EECs (option 1), and between any two SSUs, the number of EECs should not exceed 20.

NOTE 1 – Values for "option 2" clocks are for further study.

NOTE 2 – The use of such technique requires that all network elements in the synchronization chain between the primary reference clock and the network element that shall be synchronized (e.g., base station) support synchronous Ethernet.

Note that the delivery of synchronization signal at layer 1 is possible within one network operator domain. The interworking between different network operator domains is for further study.

12.2.2 Recommendations for reference timing signals distribution over packets

Packet-based methods are based on adaptive clock recovery methods, and therefore the performances are generally impacted by packet delay variation in the network (see clause 10).

Because of this, when using such techniques, it is preferable to carry timing over a well-engineered network with the timing flow carried in a channel that minimizes the packet network impairments. A part of this process may be to assign the highest priority to such a flow.

It should be noted that the shared nature of transmission implies that all flows interfere with each other to some degree regardless of priority with respect to timing.

What constitutes a well-engineered network to transport timing is currently under study. In particular, as part of this study, some metrics are being considered. These shall be based on the set of packet delays and packet delay variations.

NOTE 1 – The terms packet delay and packet delay variation (PDV) used in this Recommendation are based on definitions provided in [ITU-T Y.1540].

The characteristics of the oscillator implemented in the end equipment are among the key factors with respect to the performances that can be achieved. Therefore, the characteristics of the oscillator shall depend on the requirements to be fulfilled by the recovered timing signal, and on the level of noise (packet delay variation) generated by the network.

The specification for the clocks to be implemented in the clock recovery function of the packet based methods is outside the scope of this Recommendation.

The actual packet format is not significantly impacting the performance of packet-based methods. Proper length and basic characteristics in terms of time stamp data format shall, however, be used (PTP and NTP are two examples of time stamp format).

NOTE 2 – Methods involving support from the nodes in packet network to reduce the impact from packet delay variation (e.g., means to measure the packet delay variations that has been added by each hop in the packet network) are under study (e.g., as part of the [b-IEEE 1588] Telecom profile studies). Architectural aspects, scalability, clock requirements, etc., are among the aspects to study.

NOTE 3 – This type of approach implies to implement some hardware support in the network equipments involved in the synchronization path.

12.2.2.1 Recommendations for Deployment Case 1

The network limits for the reference timing signals in this case are defined in clause 9.2.2 for Deployment Case 1.

The requirements in this case are expressed in terms of very low levels of noise (jitter and wander) that can be generated in the network.

In this case, packet-based methods can be suitable only in case of low levels of packet delay variation generated by the network on the packets selected by the timing recovery algorithm.

The clock characteristics, metrics and related limits under which deployment Case 1 network limits can be met are under study.

12.2.2.2 Recommendations for Deployment Case 2

The network limits for reference timing signals in this case are defined in clause 9.2.2 for Deployment Case 2.

In this case, the level of noise that can be generated depends on the tolerance of the end application.

As an example, the typical wander that is tolerated by end applications can be expressed in terms of G.823 and G.824 traffic masks. Further work is needed within the standards in order to provide a general description of the conditions under which a packet network is suitable to support packet-based methods (e.g., in terms of the packet delay variation and related metrics for the packets that are used for the timing recovery algorithm), and of the clock characteristics to be used by the clock recovery algorithm.

Further information is provided in Appendix III.

Annex A

Proposed network architecture for synchronous Ethernet

(This annex forms an integral part of this Recommendation)

A.1 PRC Location

A typical synchronous Ethernet architecture will have a PRC located in one of three positions dependent on the overall architecture that the network operator wishes to follow. However, these can be summarized into three generic locations.

See Figure A.1, either:

- Case A core located – The PRC will be located at the core node, see Figure A.1 location "A". This architecture suggests a few PRC nodes, i.e., centrally located with some form of distribution to the IWF.
- Case B Access Located – The PRC will be located at some point further back within the network (geographically separate to the IWF) typically at the multi service access point, see Figure A.1 location "B". This architecture suggests a greater number of PRC nodes to that required for Case A, i.e., the PRCs are centrally located with some form of distribution to the IWF.
- Case C IWF located – The PRC will be located geographically with the IWF and that there will be a direct synchronization connection to the IWF, see Figure A.1 location "C". This suggests many PRC nodes, i.e., one PRC per IWF.

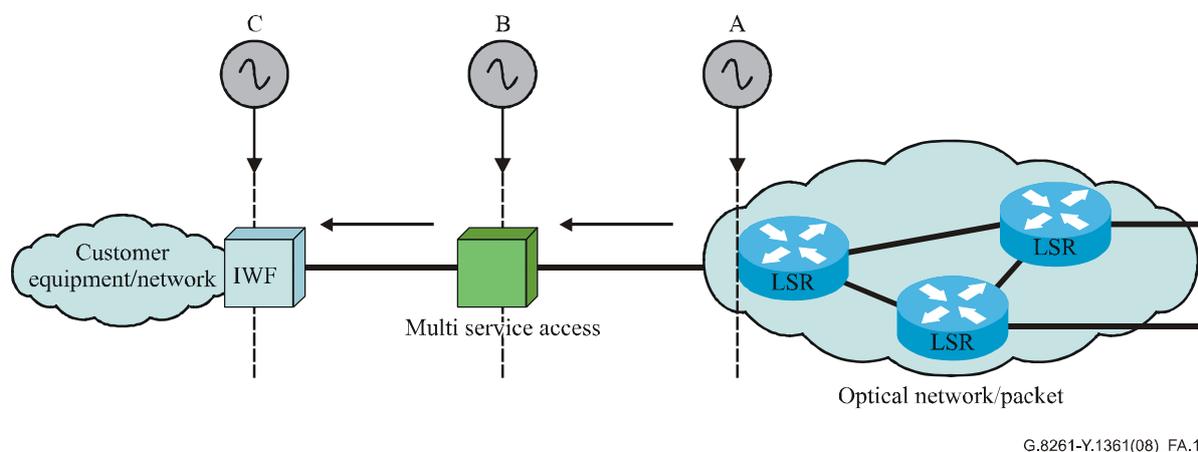


Figure A.1 – Reference clock location

Referring to the figure above, the synchronization flow is provided from the core network to the IWF. It is not intended to distribute the timing from customer equipment towards the core network.

A.2 Limiting jitter and wander of synchronous Ethernet

Limiting the jitter and wander production of the synchronous Ethernet solution in a wide area network environment is a requirement to meet the network limits.

The synchronization function within the Ethernet switch supporting synchronous Ethernet should be based upon the performance characteristics of an embedded clock. Such a clock shall be compliant with [ITU-T G.8262] in order to ensure that proper network operation occurs when such a clock is synchronized from another similar synchronous Ethernet clock or a higher quality clock. Use of such a network clock ensures synchronization interworking compliance when such a synchronous

Ethernet solution is combined with [ITU-T G.812] SSU or SASE, and hence to an [ITU-T G.811] PRC, as specified in master-slave synchronization modes of operation. It also allows interworking between existing TDM networks and new packet network architectures.

It should also be noted that this work does not impact any existing [IEEE 802.3] specifications for frequency tolerance, etc., but refers to the new additional network element clock functionality.

A.3 Considerations on the design of synchronization network based on synchronous Ethernet

The proper design of the synchronization network is the basic prerequisite to guarantee that a reference timing signal is distributed according to proper quality and reliability.

This is also applicable in case the synchronization network is based on synchronous Ethernet.

In particular, as the architecture of synchronization networks based on synchronous Ethernet may be more complex than synchronization networks based on SDH (e.g., due to meshed structure and due to potential implementation of equipment not supporting synchronous Ethernet), the synchronization planning activity is even more important.

In particular, the design shall analyse all network elements that are involved in the synchronization trail.

These shall either implement the EEC or (in case of some simple Network Elements as the 2 port devices) at least some through timing function in a similar way as done for the SDH regenerators.

The through-timing function for synchronous Ethernet equipment is for further study.

NOTE 1 – The characteristics of the clocks implemented in the equipment deployed at the end of the synchronization chain depend on the specific application implemented in the end equipment. The characteristics of these clocks may deviate from some of the EEC specifications. This is for further study.

Another important aspect to consider is related to the proper handling of the SSM in the synchronization chain.

The synchronization status message provides a mechanism for downstream EECs to determine the traceability of the synchronization distribution scheme back to the PRC or highest quality clock that is available. The synchronization function processes the SSM.

Under upstream network failure conditions, the synchronization function takes appropriate action based on the SSM and pre-set priorities and selects an alternate synchronization feed. This may be another network feed or an external feed.

The main role of SSM in synchronous Ethernet networks, similarly to the case of SDH networks, is then to support in the design of the synchronization networks in order to properly handle fault conditions.

NOTE 2 – The SSM can help in the prevention of timing loops, but still a proper planning will be mandatory to avoid timing loops.

SSM in synchronous Ethernet networks may not always be able to discover if a link is delivered from an equipment (or link) not supporting synchronous Ethernet (and that, for instance, could be inserted by mistake in a synchronization trail). As already mentioned, this shall be prevented via the proper synchronization network design activity.

The definition of more advanced SSM functions in the future (e.g., transporting information on the full synchronization path) may provide significant benefits.

When designing the synchronization network, it should be provided that the SSM is processed in every network element that can alter the timing flow (e.g., due to internal clock entering the holdover state).

It should also be avoided that timing is distributed with good quality, but SSM is not supported.

The selection process in the receiving equipment of the reference timing signal can be configured to use or not the incoming SSM.

Further details on the SSM for synchronous Ethernet are provided in [ITU-T G.8264].

NOTE 3 – In addition to the SSM, [b-ITU-T G.781] addresses other degradations, such as loss of signal, that may be used to disqualify timing references.

A.4 Example of timing distribution via synchronous Ethernet

Figure A.2 shows an example of timing distribution over the synchronous Ethernet. The timing is distributed from the PRC to the IWFs across the packet-switched network.

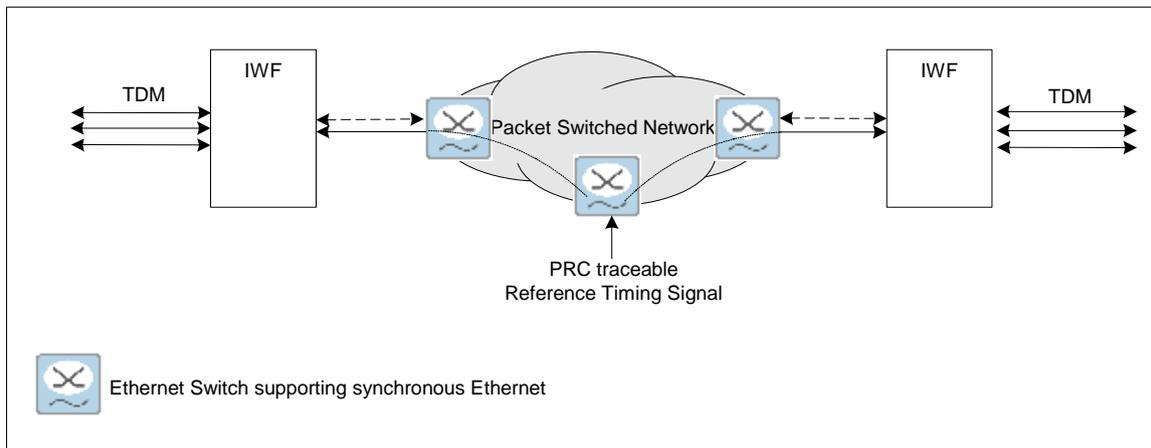


Figure A.2 – Example of timing distribution via synchronous Ethernet

A.5 Interworking of Ethernet and synchronous Ethernet interfaces

A.5.1 Interface type and operation mode definitions

The Ethernet interfaces according to [IEEE 802] are non-synchronous. With the introduction of synchronous Ethernet, it has to be distinguished between different Ethernet port types together with the different synchronization-related operation modes.

The following modes are defined (see [ITU-T G.8264]):

- Non-synchronous operation mode
- Synchronous operation mode

The default mode for a synchronous Ethernet interface is non-synchronous operation mode.

See Table A.1.

Table A.1 – Interface type and operation mode definitions

Interface type	Operation mode	QL process	ESMC process
Ethernet	Non-synchronous mode	No	No
Synchronous Ethernet	Non-synchronous mode	No	No
	Synchronous mode (QL-enabled mode)	Active, all values	Yes
	Synchronous mode (QL-disabled mode)	Inactive	Optional

A.5.2 Interworking requirements

Table A.2 shows the requirements for interworking between the different interface types and operation modes. Any combination must enable the proper transmission of the Ethernet traffic. To use synchronous Ethernet for network synchronization, synchronous Ethernet ports in synchronous mode are required at both ends of each synchronization link involved in the synchronization path.

Table A.2 – Interworking between synchronous Ethernet and Ethernet ports

Interface type		Traffic interworking with			Network synchronization interworking with		
		Ethernet	Synchronous Ethernet port		Ethernet	Synchronous Ethernet port	
			Non-synchronous mode	Synchronous mode		Non-synchronous mode	Synchronous mode
Ethernet	Non-synchronous mode	✓	✓	✓	✗	✗	✗
Synchronous Ethernet	Non-synchronous mode	✓	✓	✓	✗	✗	✗
Synchronous Ethernet	Synchronous mode	✓	✓	✓	✗	✗	✓
✓ Interworking is possible ✗ Interworking is not possible							

A.5.3 Interworking in frequency

Ethernet works with ± 100 ppm as maximum frequency offset.

Synchronous Ethernet in synchronization mode uses network synchronization derived from PRC. The worst case is holdover with ± 4.6 ppm as maximum frequency offset.

For data recovery, the synchronous Ethernet frequency input tolerance has to be ± 100 ppm.

Table A.3 shows the requirements for interworking in terms of frequency.

Table A.3 – Interworking in frequency

Interface type	Operation mode	Frequency		
		Maximum output frequency deviation	Input tolerance	
			For data recovery	For clock recovery
Ethernet	Non-synchronous mode	±100 ppm	±100 ppm	n/a
Synchronous Ethernet		It might be locked to the EEC or, if not, be within ±100 ppm		
Synchronous Ethernet	Synchronous mode (Note)	Locked to the EEC (in the worst case ±4.6 ppm)		Max. ±4.6 ppm

NOTE – For QL-enabled and QL-disabled modes.

A.5.4 Interworking in noise

Ethernet specifies jitter according to IEEE. Wander is not an issue for Ethernet traffic operation.

Jitter and wander for synchronous interfaces is specified according to ITU-T. For Synchronous Ethernet interfaces in synchronous operation mode, the relevant requirements are specified in Recommendation ITU-T G.8261 and [ITU-T G.8262].

For data recovery, a synchronous Ethernet interface has to be able to tolerate the jitter from Ethernet interfaces.

Table A.4 shows the details.

Table A.4 – Interworking in noise

Interface type	Operation mode	Noise					
		Maximum output noise generation		Equipment input noise tolerance			
		Jitter	Wander	For data recovery		For clock recovery	
				Jitter	Wander	Jitter	Wander
Ethernet	Non-synchronous mode	According to IEEE	n/a	According to IEEE	n/a	n/a	n/a
Synchronous Ethernet							
Synchronous Ethernet	Synchronous mode	According to G.8261 (Network), G.8262 (Equipment)				According to G.8262	

A.5.5 Related jitter measurement

Jitter measurements of Ethernet ports refer to IEEE. The IEEE jitter measurement uses the method stressed eye and bath tub diagram. For data recovery interworking, the synchronous Ethernet interfaces must be measured in the same way.

For synchronization interworking of synchronous Ethernet interfaces in synchronization operation mode, the jitter specifications are included in [ITU-T G.8261] and [ITU-T G.8262].

Jitter measurements for synchronous Ethernet interfaces in synchronization operation mode are for further study. See Appendix X.

Table A.5 provides a summary.

Table A.5 – Jitter measurement

Interface type	Operation mode	Jitter input tolerance	Jitter noise generation	Jitter noise transfer	Network limits
Ethernet	Non-sync mode	According to IEEE	According to IEEE	n/a	n/a
Synchronous Ethernet					
Synchronous Ethernet	Sync mode	For further study, see Appendix X for jitter measurements			

A.5.6 Related wander measurement

Wander requirements are not specified for Ethernet interfaces.

Wander measurements for synchronous Ethernet interfaces in synchronization operation mode are for further study. See Appendix X.

See Table A.6 for details.

Table A.6 – Wander measurement

Interface type	Operation mode	Wander input tolerance	Wander noise generation	Wander noise transfer	Network limits
Ethernet	Non-sync mode	n/a			
Synchronous Ethernet					
Synchronous Ethernet	Sync mode	For further study; see Appendix X for jitter measurements			

NOTE – Considerations on interfaces and equipment with reduced synchronous Ethernet functionalities are provided in [ITU-T G.8264].

Annex B

IWF functional partitioning into CES and PNT IWF and network examples

(This annex forms an integral part of this Recommendation)

B.1 General

The IWF is the functional block that translates data from a TDM-based network towards a packet-based network, and vice versa (see Figure B.1).

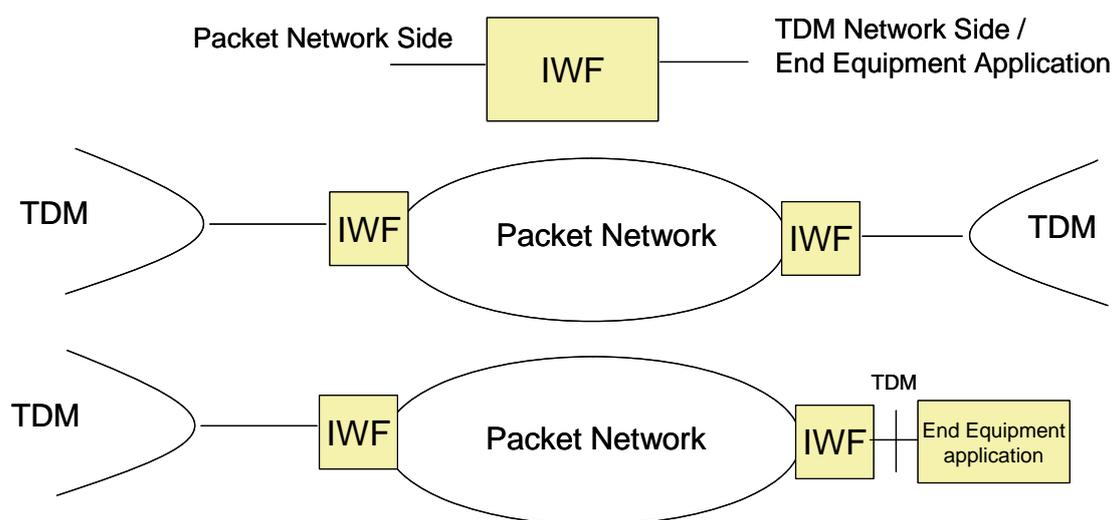


Figure B.1 – IWF in the network

In some applications, the function in the IWF may change the layer over which the timing is carried (i.e., from packet to physical and vice versa), see Figure B.2.

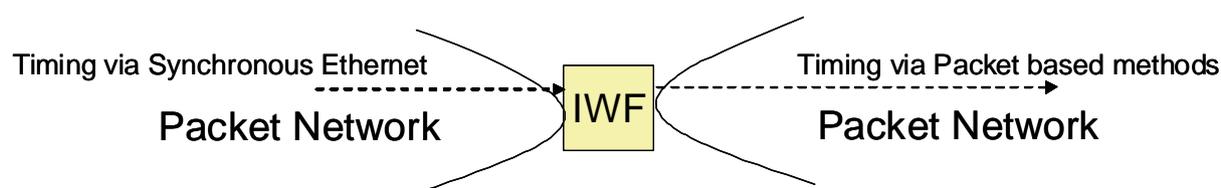


Figure B.2 – IWF changing the layer over which the timing is carried

The IWF is defined according to a two-domain structure (see Figure B.3), with a "CES IWF" taking care of the synchronization aspects of the service clock domain, and a "PNT (packet network timing) IWF" taking care of the synchronization aspects of the synchronization network clock domain.

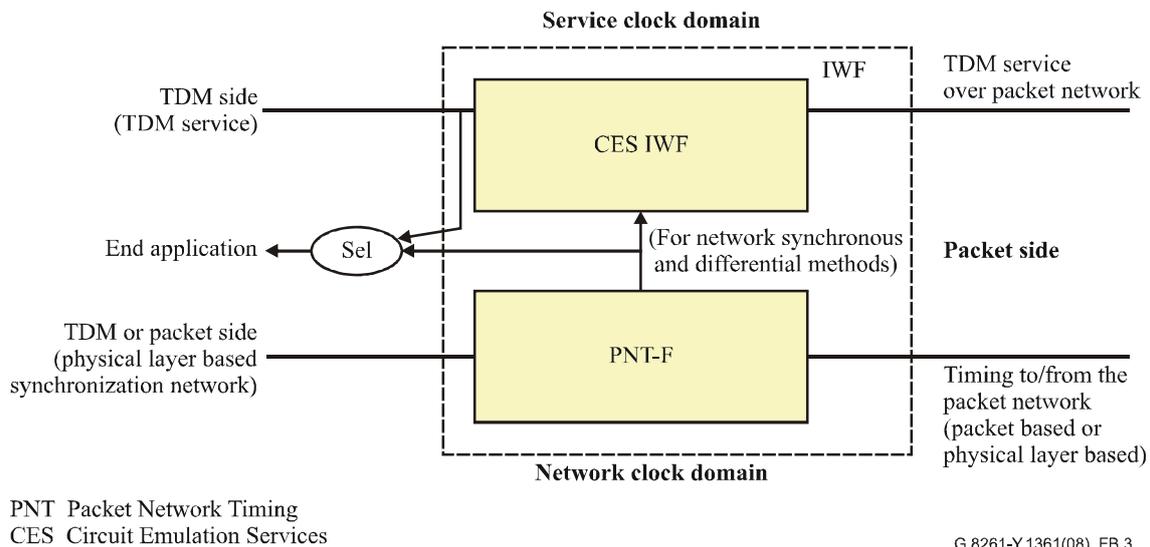


Figure B.3 – CES and PNT domains in the IWF

The CES IWF, in particular, shall recover the timing of the services that are carried over the packet network (service clock signal recovery), as well as properly support the generation of the CES packets towards the packet network.

As described in clause 8, the following operating methods are possible:

- Network synchronous operation
- Differential methods
- Adaptive methods
- Reference clock available at the TDM end systems

Depending on the applicable methods, different types of clocks may be needed to be implemented in the CES IWF.

This is detailed in clause B.3.

The timing distribution to/from the PNT IWF can be done according to traditional or new methods. In fact, this block may recover the synchronization network timing either from the TDM side (e.g., from SDH), or from the packet side (e.g., synchronous Ethernet, or new methods based on dedicated packets).

The following list is an example of the possible cases for timing distribution to/from the PNT:

- Dedicated external reference (e.g., from SASE)
- Via synchronous physical layer (e.g., SDH, synchronous Ethernet). Timing carried by packets (e.g., [b-IEEE 1588], NTP).

In some cases, the PNT IWF shall deliver to the CES IWF an accurate reference timing signal derived from the synchronization network. In fact, the CES IWF may need this reference to support the clock recovery mechanism: this applies to the network synchronous and to the differential methods. More details are provided in clause B.2.

B.2 IWF clocks

Figure B.4 shows the clocks that could be implemented in the CES IWF. These are:

- PSC-A: clock that recovers the CES service clock signal from traffic packets according to an adaptive method

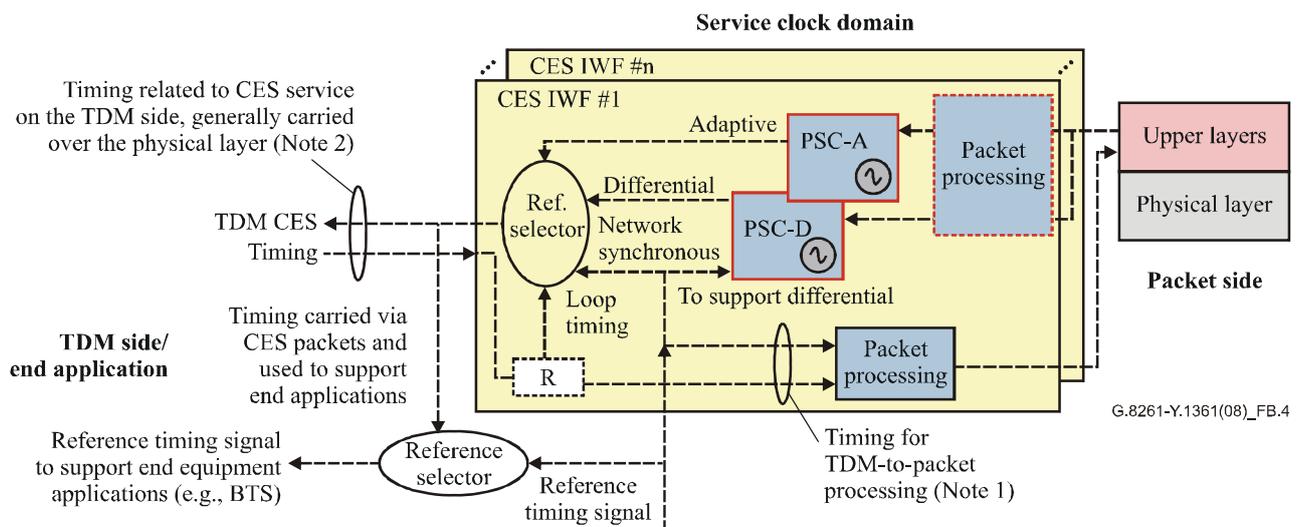
- PSC-D: clock that recovers the CES service clock signal from traffic packets according to a differential method

These clocks are responsible for the CES IWF timing function (including free running, holdover, filtering, selection, etc.).

Clocks PSC-A and PSC-D are specific per each CES user (i.e., any user has a dedicated clock). The holdover should not be mandatory (but optionally it could be provided) as, in fact, only some free running accuracy will be needed to support the PDH minimum requirements (e.g., 50 ppm for 2048 kbit/s signals). They normally handle only one-way packets to support TDM CES.

NOTE 1 – New "services" distributing dedicated timing packets could be defined (using one-way or two-way protocols) and different clocks would be defined. This is for further study.

The details of these clocks are for further study.



n Number of CES users terminated by the CES IWF

NOTE 1 – This timing drives the generation of the outgoing packets and forms the basis for generating timing messages for adaptive or differential methods.

NOTE 2 – Some of the CES flows may also carry the network clock. In case the IWF is handled by the network clock owner, these could be handled in the network clock domain (by the PEC). In case the IWF is owned by a different operator (offering the transport service to another operator), these are handled as any other TDM service in the service domain (by the PSC).

Figure B.4 – Clocks in the CES IWF and PNT IWF

From a synchronization perspective, the CES IWF on the TDM side is mainly requested to generate a synchronous physical layer according to defined requirements (e.g., jitter and wander limits), and to recover the timing according to the applicable jitter and wander tolerance masks.

On the CES IWF packet side, the synchronization function concerns the "user data" packets. For instance, in the packet-to-TDM direction and in case of adaptive methods, the TDM timing will be recovered by some filtering algorithm based on the inter-arrival time of the packets (PSC-A is taking care of this function).

Block "R" in the CES IWF is used to regenerate the TDM timing: this can be looped back in case of loop timing function, and can be used to control the packet flow on the TDM-to-packet direction (e.g., to generate the timing messages used by the differential method). In particular, the block "Packet Processing" is responsible for the generation of the timing messages supporting the differential method (both the network clock and TDM timing are needed for this purpose), and for the generation of the packets with a rate directly related to the TDM timing (valid for all methods).

Figure B.5 shows the clocks in the PNT-F which support the reference timing distribution over packet networks (see clause 7).

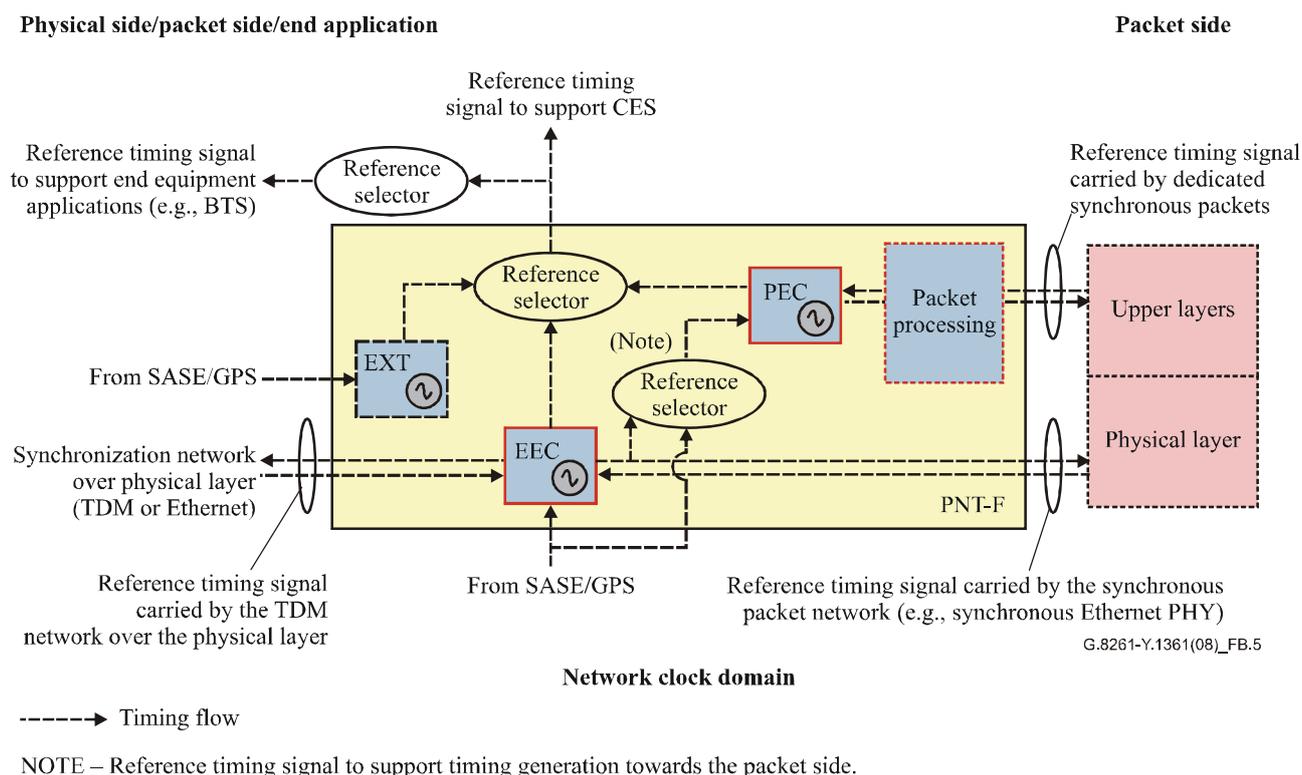


Figure B.5 – Clocks in the PNT-F

NOTE 2 – Not all the interactions between the blocks into the PNT are shown in the figure.

NOTE 3 – A different clock may be implemented in the PNT in case the timing is carried by TDM only (e.g., SDH). This clock shall be based on the applicable Recommendations (e.g., [ITU-T G.813]).

According to Figure B.5, the following PNT clocks (synchronization network clocks) are defined:

- PEC: clock to recover and send the network timing via dedicated packets
- EEC: clock to support the timing carried via Synchronous Ethernet (see [ITU-T G.8262])
- EXT: timing from an external dedicated reference timing signal (e.g., SASE/GPS)

The details on the PEC are for further study.

PEC and EEC handle the network clock, and therefore one frequency per network. There is one clock/frequency per equipment. Holdover is requested for these clocks.

The PEC may be able to handle two-way packets (e.g., PTP, NTP).

It can be noticed that the PSC-A and the PEC may be based on a similar implementation as both are based on an adaptive method, but different requirements may apply. In addition, while the PSC-A should only terminate the timing distribution (it takes care of the TDM service timing recovery for further generation of the TDM flow), the PEC may in principle be part of a synchronization distribution chain.

The clock EXT may be based on other relevant ITU-T Recommendations (e.g., [ITU-T G.812], and [ITU-T G.813]).

It shall be noted that depending on the network application, only some of the functions (and only some of the clocks) shown in Figures B.4 and B.5 have to be implemented in the IWF. As an example, clock EXT is not needed if in the PNT there are other clocks implemented (e.g., EEC): they could take care of receiving the external reference for use in the system instead.

As shown in Figure B.5, a reference timing signal can also be made available for supporting end equipment applications (e.g., BTS).

The outgoing reference timing signal shown in Figure B.4 could also be carried by a CES signal. The recovered timing signal could be made available as an external interface (e.g., [ITU-T G.703] compliant) or could be internal to the system (in this case, the IWF is integrated into the end application using this timing reference).

Examples of typical IWF applications are provided in Appendix IX.

Examples on the network applications are described in clause B.3.

NOTE 4 – In addition to the IWF clocks, there can be clocks implemented in other network elements of the packet network. A typical example is the EEC in the Ethernet switches which are part of the synchronization distribution chain. In this case, the network element implements only PNT functions and the synchronization functions can be represented by the applicable clock (e.g., EEC).

More generally, in case there are no CES functions and the EEC is connected on both sides to the same type of physical layer (e.g., synchronous Ethernet), there is no IWF.

B.3 Network examples

Some network examples are shown in the following figures in order to get a better understanding of the model presented in Figures B.1 and B.2.

Figure B.6 shows the TDM service clock recovery based on a network synchronous method: in this example, the reference timing signal (network clock) is distributed from the PNT IWF having access to a PRC (e.g., GPS), towards the PNT/CES IWF at the edges of the packet network (e.g., via synchronous Ethernet). The TDM service timing is derived from this network clock (i.e., network synchronous method).

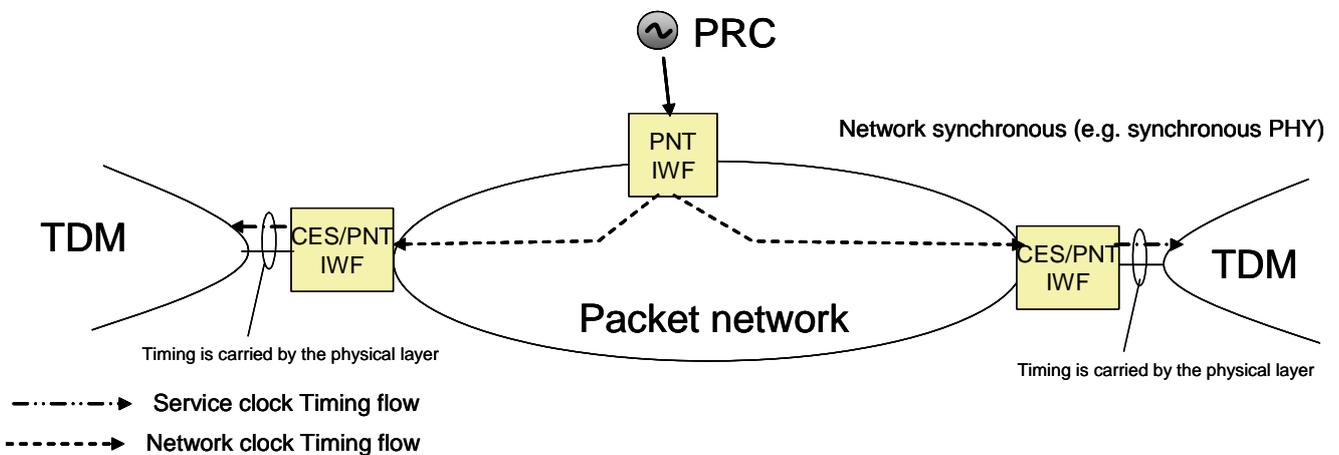


Figure B.6 – Example on timing recovery based on network synchronous method

Figure B.7 presents the service clock recovery based on the differential method. In this example, the PNT IWF distributes the reference timing signal to the CES/PNT IWF that will use this signal to implement a differential method (the flow from the left side IWF to the right side IWF represents the distribution of the differential information via timing messages).

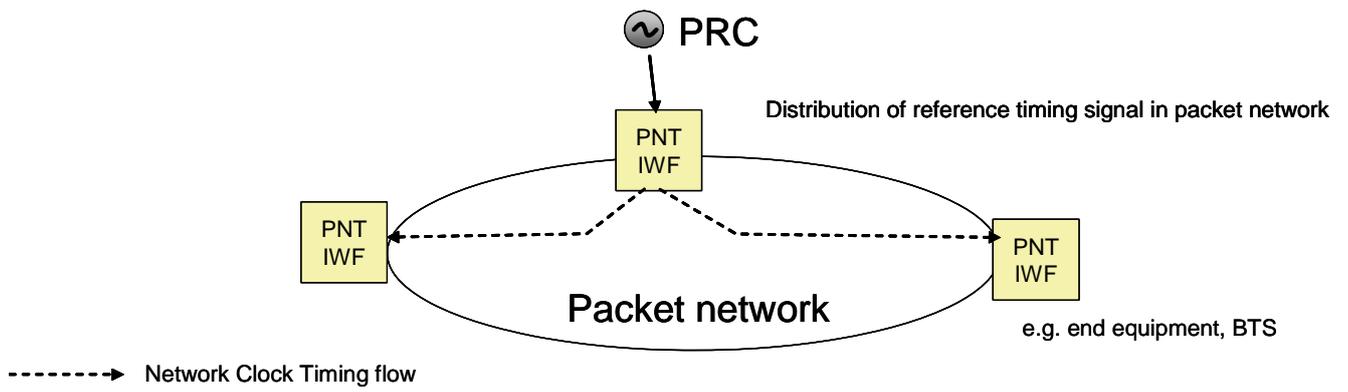


Figure B.9 – Reference timing distribution between IWF PNT

Annex C

CES IWF synchronization related requirements

(This annex forms an integral part of this Recommendation)

C.1 Traffic interfaces

The following requirements have been taken from existing Recommendations (e.g., [ITU-T G.823], [ITU-T G.824], etc.).

NOTE – The SDH interfaces are mentioned in the following clauses only for information purposes as the transport of the SDH signals over packet network is for further study.

C.1.1 Physical, electrical and optical characteristics

Physical and electrical characteristics of E0 (64 kbit/s), E11 (1544 kbit/s), E12 (2048 kbit/s) interfaces, all PDH interfaces, 51 840 kbit/s (STM-0) and ES1 (STM-1) interfaces shall comply with the requirements of [ITU-T G.703].

Physical and optical characteristics of STM-1, STM-4, STM-16 interfaces shall comply with requirements of the relevant physical interface Recommendations e.g., [ITU-T G.957], [ITU-T G.691], [ITU-T G.959.1], etc.

C.1.2 Jitter and wander tolerance

The input jitter and wander tolerance for networks based on 2048 kbit/s hierarchy at E0, E12, E22, E31, E4 traffic interfaces shall comply with requirements of clause 7.1 of [ITU-T G.823].

The input jitter and wander tolerance for networks based on 1544 kbit/s hierarchy – at E11, E21, 32 064 kbit/s, E32, 97 728 kbit/s traffic interfaces shall comply with requirements of clause 7.2 of [ITU-T G.824].

The input jitter tolerance for SDH-based networks – at STM-1e, STM-1, STM-4, STM-16 traffic interfaces shall comply with requirements of clause 6.1.2 of [ITU-T G.825]. The input jitter tolerance at 51 840 kbit/s traffic interface shall comply with requirements of clause 16.3 of [ITU-T G.703].

The input wander tolerance for SDH-based networks – at 51 840 kbit/s, STM-1e, STM-1, STM-4, STM-16 traffic interfaces, according to clause 6.1.1 of [ITU-T G.825], shall comply with requirements of clause 9.1 of [ITU-T G.812] and clause 8.1 of [ITU-T G.813], whichever is applicable. These requirements are defined for synchronization interfaces (SSU and SEC respectively) because STM-N traffic interfaces are considered as synchronization interfaces.

Measurement methods are defined in [ITU-T O.171] and [ITU-T O.172].

C.2 Synchronization interfaces

The following requirements have been taken from existing Recommendations (e.g., [ITU-T G.703], etc.).

C.2.1 Physical and electrical characteristics

Physical and electrical characteristics of T12 (2048 kHz) synchronization interface shall comply with requirements of clause 13 of [ITU-T G.703].

Physical and electrical characteristics of E12 (2048 kbit/s) synchronization interface shall comply with requirements of clause 9 of [ITU-T G.703].

Physical and electrical characteristics of E11 (1544 kbit/s) synchronization interface shall comply with requirements of clause 5 [ITU-T G.703].

C.2.2 Jitter and wander tolerance

The input jitter tolerance at T12, E12 synchronization interfaces, according to clause 7.2 of [ITU-T G.823], shall comply with requirements of clause 9.2, Type I of [ITU-T G.812] for SSU interfaces and clause 8.2, Option 1 of [ITU-T G.813] for SEC interfaces, whichever is applicable.

The input jitter tolerance at E11 synchronization interface, according to clause 7.3 of [ITU-T G.824], shall comply with requirements of clause 9.2, Types II and III of [ITU-T G.812] for SSU interfaces and clause 8.2, Option 2 of [ITU-T G.813] for SEC interfaces, whichever is applicable.

The input wander tolerance at T12, E12 synchronization interfaces, according to clause 7.2 of [ITU-T G.823], shall comply with requirements of clause 9.1, Type I of [ITU-T G.812] for SSU interfaces and clause 8.1, Option 1 of [ITU-T G.813] for SEC interfaces, whichever is applicable.

The input wander tolerance at E11 synchronization interface, according to clause 7.3 of [ITU-T G.824], shall comply with requirements of clause 9.1, Types II and III of [ITU-T G.812] for SSU interfaces and clause 8.1, Option 2 of [ITU-T G.813] for SEC interfaces, whichever is applicable.

C.3 IWF synchronization function

Details of the IWF synchronization function are provided in Annex B.

Depending on the services to be provided, a suitable subset of the timing function described in Annex B shall be supported by the IWF.

With reference to the CES IWF in Figure B.4, it is recommended to have slip control in the TDM Tx direction to control possible over/underflow in the buffer. Slips shall be performed on $n \times 125$ microsecond frames.

When TDM transmitter and/or receiver clocks are in holdover or are traceable to clocks in holdover, and a synchronous clock recovery technique (differential method or network synchronous operation) is used, slips (most likely uncontrolled) will occur.

When selecting a new timing source, the output wander may temporarily exceed the output wander limit. However, the output wander must be within the output wander limit by the end of a period called the "Stabilization Period". The requirements for the stabilization period are for further study; more information is provided in Appendix II.

Another characteristic that is relevant for the IWF is the latency. The latency requirements are normally defined on the network level specifying the total latency in the end-to-end connection. Requirements on IWF contribution on the total latency is for further study.

The specification of the total noise that can be introduced by a CES segment is defined in clause 9.1. This also defines the noise transfer characteristics of the total CES segment, including the IWFs pair that adapts the TDM flow into the packet network.

Annex D

Network applications and requirements for clocks specified in G.8262/Y.1362

(This annex forms an integral part of this Recommendation)

A synchronization network according to [ITU-T G.803] uses PRC(s), SSUs and SECs. The SSUs are often stand-alone equipment. The timing information is transferred via SDH network elements (NEs) from a PRC to an SSU and from an SSU to an SSU of a lower hierarchical level. Two or more routes are used for resilience. This is illustrated in Figure D.1-a.

With the introduction of synchronization in packet-switched networks (PSN), the packet-switched NEs supporting synchronous Ethernet must be able to transfer timing information and interwork with SDH NEs (e.g., containing SEC). The packet-switched network elements containing EEC must be able to provide synchronization lines between PRC and SSUs and supply synchronization to time sensitive applications. The new timing links via packet-switched networks must be in line with existing SDH timing links for inter-operability with the synchronization network. Figure D.1-b shows two synchronization chains, one formed by SDH NE (circles with "S"), and the other formed by packet-switched NEs using synchronous Ethernet interfaces (circles with E).

Hybrid NEs are described in Appendix I of [ITU-T G.8262]. Hybrid NEs that offer both STM-N interfaces with associated SDH-VC cross-connect functions and synchronous Ethernet interfaces (ETY) with associated packet switching. It should be possible to use such hybrid NEs at any place in synchronization chains. An example is illustrated in Figure D.1-c. The upper hybrid NE (circle with H) uses an STM-N interface at the ingress and an ETY interface at the egress. The lower hybrid NE uses an ETY interface at the ingress and an STM-N interface at the egress. Timing is transferred from STM-N to ETY and from ETY to STM-N, respectively.

The clock characteristics of EECs may support the construction of timing distribution chains providing the same behaviour as chains of SECs (see Figure D.1-b).

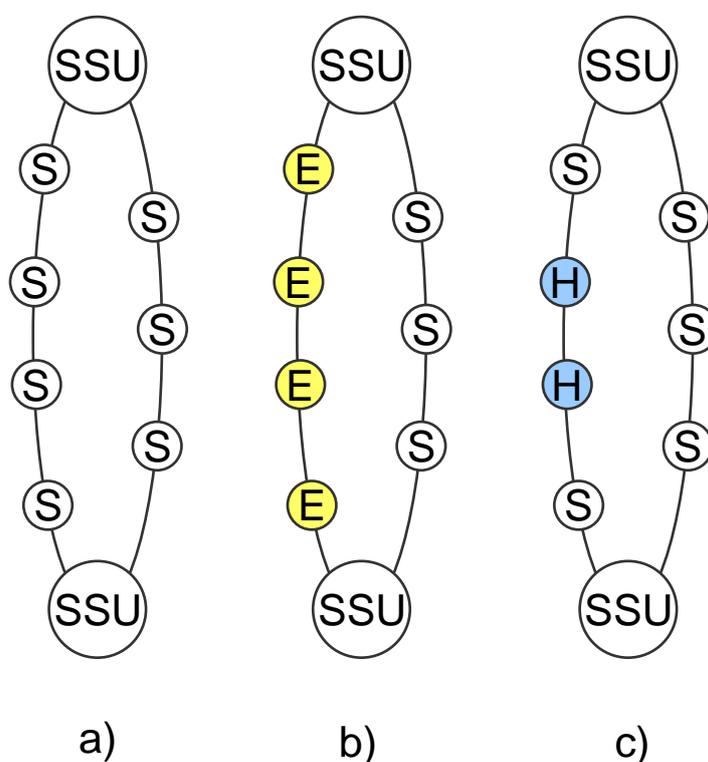


Figure D.1 – Synchronization chains implemented with different types of NEs

Appendix I

Characteristics of Ethernet switches and networks

(This appendix does not form an integral part of this Recommendation)

I.1 Delay characteristics of Ethernet switches

I.1.1 Functional operations within an Ethernet switch

From a "black box" perspective, an Ethernet frame passes through four functional operations in a typical Ethernet switch. These are shown in Figure I.1:

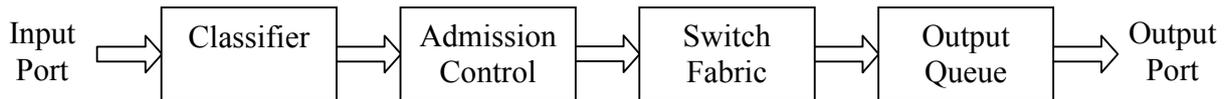


Figure I.1 – Typical functions within an Ethernet switch

- Classification – Identification of the flow to which the frame belongs, and determination of the output port and priority
- Admission control – Application of traffic management for the flow (policing, shaping, marking)
- Switching – Forwarding to the appropriate output port
- Output queue – Waiting for a transmission slot on the output port. Typically, queuing policies, such as strict priority, weighted fair queuing or round robin are applied.

The following clauses discuss the delay properties of the various functions within a switch.

I.1.2 Input stage delay

The time required for the classification and admission control stages should be approximately constant in most cases. However, depending on the switch design and the traffic loading on the switch, the delay through these functions may vary. For example, in some switches, both classification and admission control may be performed in software on a network processor. At full load, the software may not be able to keep up with the number of frames to be processed, hence the delay may increase, and some frame dropping may occur. The same may also be true of some hardware-based designs.

Figure I.2 shows a simplistic view of the variation of input stage delay with switch loading. Under low traffic loads, the switch can cope with the number of frames passing through it without adding to the delay. As the frame rate increases, while the overall processing capacity of the switch is not exceeded, the instantaneous frame rate may exceed the available processing rate. This will cause frames to be buffered awaiting processing, and some extra delay to be incurred. Finally, at some point, the mean incoming frame rate may exceed the processing capacity, causing the delay to be increased further, and, in some cases, frames to be dropped due to lack of buffering capacity.

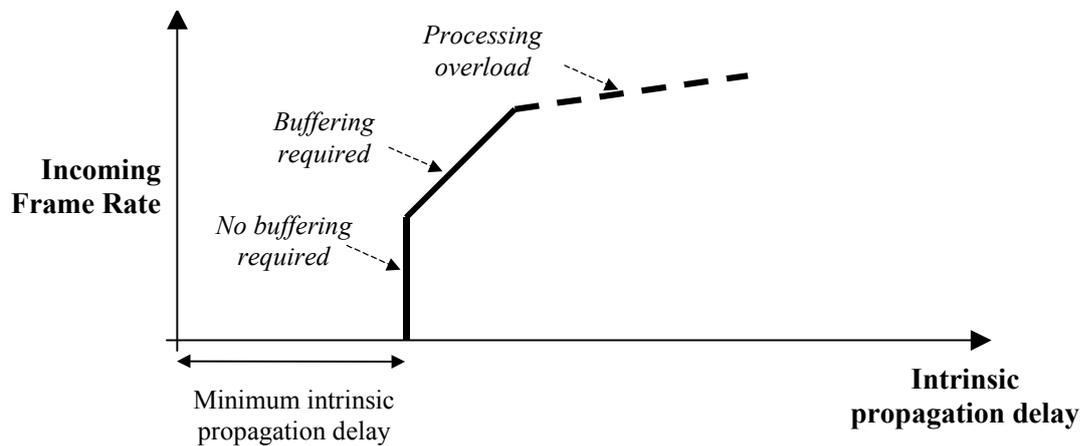


Figure I.2 – Variation of input stage delay with loading

I.1.3 Switch fabric delay

The delay through the switch fabric itself is also dependent on both switch architecture and traffic loading. For example, many switches operate scheduling algorithms for switching of frames from input ports through to output ports, and this may cause a small variation in delay to the frames, depending on their arrival time relative to the scheduler "tick". However, in most cases, this variation in delay is small due to the high frequency at which the scheduler works.

At very high incoming data rates, the switch fabric itself may be overloaded, and unable to cope with the full volume of traffic requiring switching. This will result in frames being dropped.

I.1.4 Output queuing delay

The amount of delay added by the output queue will depend on the queuing policy employed, and the priority of the traffic flow. For example, a high priority flow (such as might be used for a packet timing flow) in conjunction with a strict priority policy might experience "head-of-line blocking" delay. This is where although a frame has highest priority, it arrives at the output port just after a low-priority frame has started to be transmitted. The high priority frame then has to wait until the other frame has finished transmitting.

Figure I.3 shows the delay profile experienced by a population of high priority frames in conjunction with a strict priority queuing policy. For the purposes of simplicity, this diagram assumes frames experience an approximately constant delay through the other switch functions, here termed "intrinsic propagation delay through switch". A proportion of frames arrive at the output queue at a time when there are no other frames currently being transmitted. These frames are transmitted immediately. The remainder has to wait in the queue while the current transmission completes. There may also be an additional delay, due to other high priority packets also in the queue.

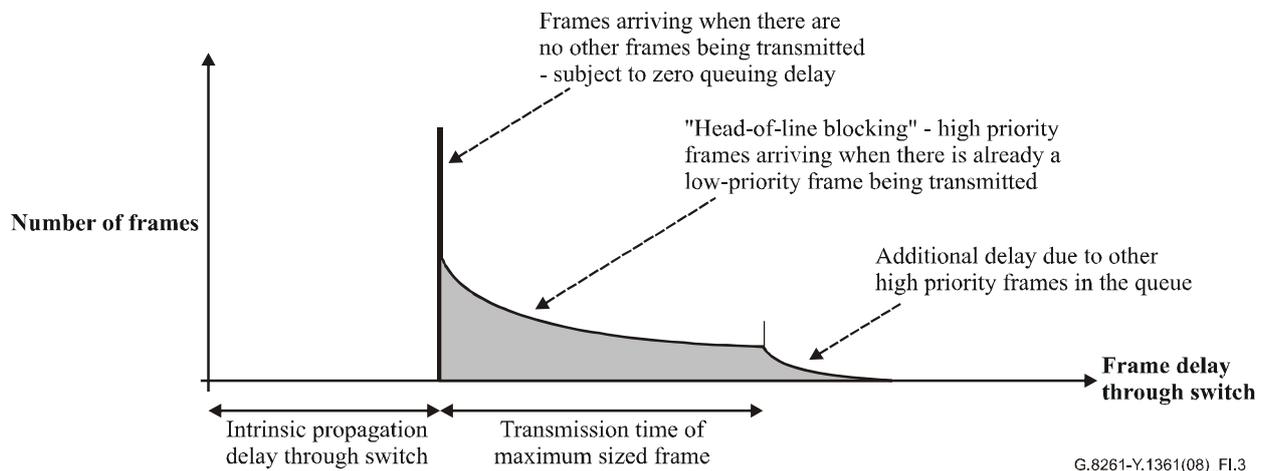


Figure I.3 – Strict priority queuing: Head of line blocking

I.1.5 Typical delays in the Ethernet switches

Based on the model described in clause I.1, it is possible to provide a simplified modelling of the delays caused by an Ethernet switch, identifying two main contributions.

The first type of contributions is related to the classification, admission control and switching operations; the second type of contribution is related to the output queue and transmission.

The first type of the delay is then mainly related to the processing capacity of the switch, while the second one depends on the bit rate of the outgoing line (e.g., 1 Gbit/s) and on the queuing policy/priorities that are implemented.

Assuming that the design of the Ethernet network will not implement Ethernet switches where the bottleneck is the processing capacity of the Ethernet switch, one could assume that the processing capacity should contribute with values below 10 microseconds (in fact a 1500 bytes packet in the output queue takes 12 microseconds on 1 Gbit/s link), and, in addition, the processing overload or processing buffering should not be an issue (see Figure I.2).

With respect to the second type of delay, these can be calculated according to the model provided in Appendix V.

The simplified model is shown in Figure I.4.

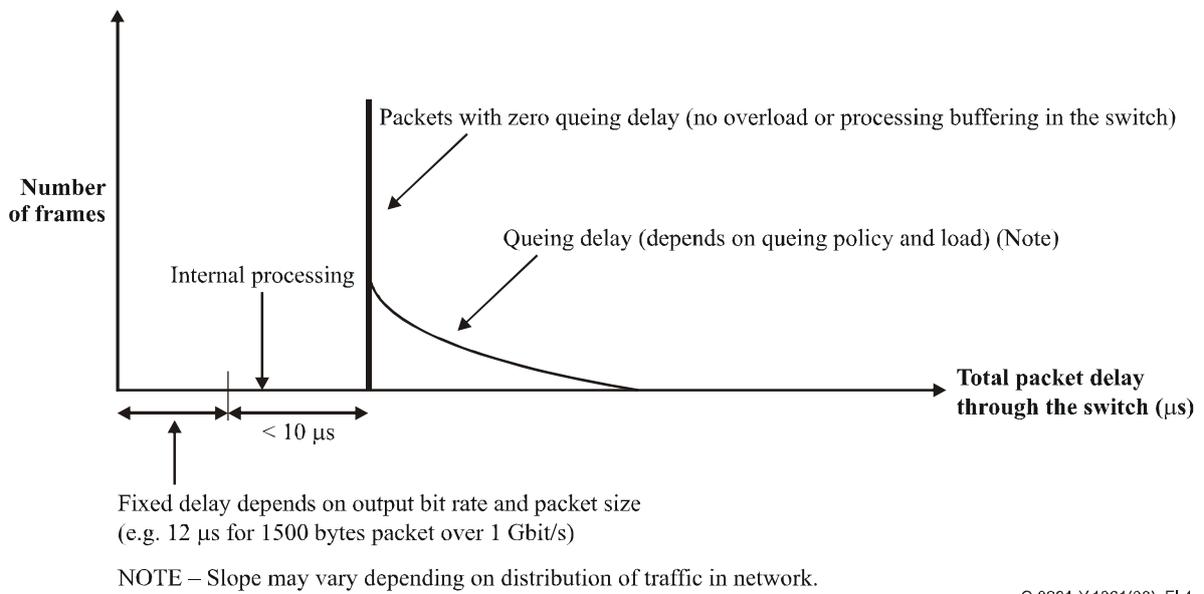


Figure I.4 – Simplified model of the delays in the Ethernet switch

Referring to Figure I.4, it shall be noted that the queue processing may also impact the shape of the delay distribution.

I.2 Characteristics of switched Ethernet networks

I.2.1 Topology of Ethernet networks

Although there are many different possible network topologies, for the purpose of considering a particular flow through a network, it can be modelled as a chain of Ethernet switches as shown in Figure I.5. At each switch in the chain, an Ethernet frame has the potential to be delayed due to the mechanisms described in clause I.1. This delay will be affected by the other traffic flowing through the switch. Traffic directed to the same output port will affect the output queuing delay, while the sum total of all traffic flowing through the switch (including that flowing to other ports) will affect the processing and switch fabric delays.

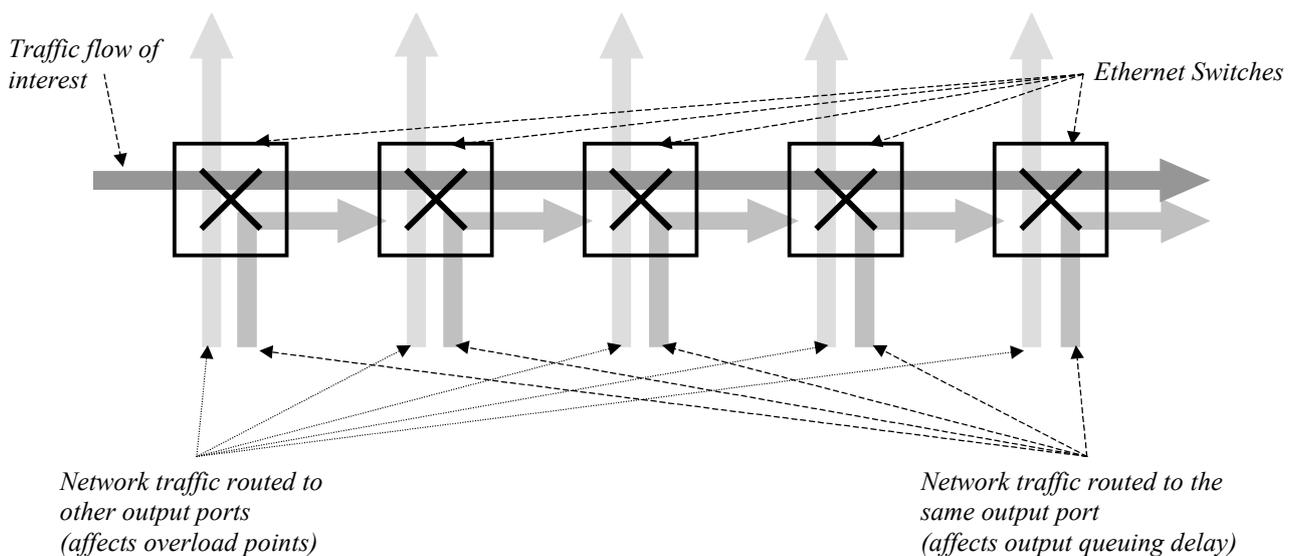


Figure I.5 – Data flows within an Ethernet network

The length of the chain affects the total delay of the system; clearly the more switches there are, the greater the total delay, also the greater the delay variation. However, in many Ethernet networks, the length of the chain may be quite small. For example, in a hierarchical network, there may often be only two or three levels of hierarchy, yielding a chain length of up to five switches.

In some instances, a ring topology may be employed. Typically, these may contain around ten switches, giving a maximum "distance" around the ring of five switches. Occasionally, interconnected rings may be used, which could double the "distance" to around ten.

I.2.2 Traffic patterns and levels

With the exception of constant bit rate and real-time traffic, most network traffic is extremely bursty in nature. It has been observed that on almost any level one cares to look, traffic variation can be observed. For example, at a very small level there is bursting due to the opening and closing of the TCP window size. At a larger level, there may be bursting due to the nature of the application (e.g., downloads of large files), while at a larger level still there may be bursting due to the time of day (e.g., higher activity levels during the day than at night).

When considering the delay performance of a TDM transport flow, the effects of other traffic in the network have to be considered. For example, in Figure I.5, each of the network traffic flows may be varying in some form, independently of the other flows.

[b-ITU-T G.1020] proposes the use of four-state Markov models for modelling packet loss distribution. A similar technique could be applied to burst lengths in each flow, allowing bursts and groups of bursts to be modelled. The longer term (e.g., diurnal variation) could then be applied as a gradual variation in burst densities.

I.2.3 Disruptive events in Ethernet networks

There are several types of "disruptive events" that may cause sudden changes in delay in an Ethernet network. The resulting delay changes may be permanent or temporary. These disruptive events include:

- Routing change, causing a permanent step change in delay
- Temporary network overload, causing a significant but temporary delay change
- Temporary loss of service, causing all packets to be lost for a period.

Appendix II

Stabilization period

(This appendix does not form an integral part of this Recommendation)

The stabilization period is a parameter that may be important during the start-up phase (in order to get a quick installation of the equipment), or when switching between timing references (in order to limit the phase transient). In case the equipment has been operating in holdover for long periods (e.g., hours), the phase error when selecting a new clock reference would be largely dominated by the phase error caused by the frequency error of the clock in holdover.

In case the adaptive method is used, the requirement on the stabilization period may depend on the actual phase noise in the packet network. In fact, a high packet delay variation in the packet network may require a long period before the clock can lock to the timing reference.

The filter implementation and the characteristics of the internal oscillator are important as well. In fact, depending on the holdover characteristics (e.g., [ITU-T G.812] Type II vs Type III), longer time could be accepted when switching from a reference to a second reference, as a good holdover can allow longer locking periods (the main requirement is to limit the total phase error during reference switching).

The requirements on the stabilization period are under study.

For the purpose of the tests detailed in Appendix VI, a stabilization period of at least 900 s is proposed for the adaptive methods as, in order to properly characterize the packet delay variation statistics in a network, a sufficiently long period might be required.

Appendix III

Considerations on packet-based methods

(This appendix does not form an integral part of this Recommendation)

In some applications, it is required to recover reference timing signals that comply with wander that can be expressed in terms of G.823 and G.824 traffic masks. Under certain circumstances, these network limits are within the achievable performances of the packet-based methods in packet networks properly designed (e.g., networks that can be modelled as network presented in Model A, see Appendix V). What constitutes a packet networks properly designed is currently under study, see also clause 12.2.2.

For the case of mobile backhauling, the use of packet-based method to synchronize the BTS/Node B will depend on a number of complex issues significantly determined by the embedded functionality of the BTS/Node B.

The following needs to be considered:

- 1) Stability of the oscillator in the BTS/Node_B
- 2) Physical layer interface to the BTS/Node_B (e.g., TDM or Ethernet)
- 3) Tolerance specification at the input of the BTS/Node_B (specified in terms of G.823/G.824 traffic interface masks by 3GPP in case of TDM interfaces).

With respect to point 1), when frequency accuracy is the only concern, a stable oscillator could allow to relax the requirements on the packet delay variation in the network as longer filtering period could be designed. It shall be noted that stable oscillators are normally implemented in the base stations, due to short-term stability required on the radio interface and holdover requirements. This area is for further study (e.g., time to meet a specific requirement after the oscillator has been powered on).

Appendix IV

Applications and use cases

(This appendix does not form an integral part of this Recommendation)

IV.1 Background

The purpose of this appendix is to provide some explanatory information related to three use case categories. Special consideration is given to situations where the transport network supporting the use-case is changed from PDH/SDH to Ethernet.

There are three principal types of synchronization that are of importance. Each particular application may have different needs, and it is necessary to ensure that the transport network is capable of providing this functionality or the network operator must provide for alternate methods. The three synchronization categories are:

- a) Frequency synchronization
- b) Phase synchronization
- c) Time synchronization

Frequency synchronization relates to the alignment of clocks in frequency, a process that is also referred to as syntonization. Phase synchronization implies that the two clocks are aligned in phase, a process that is also referred to as relative-time synchronization. Time synchronization is also referred to as time-of-day synchronization or wall-clock synchronization, where the clocks in question are traceable to a common, universal, time-base such as UTC. Note that if two clocks are synchronized in time/phase, then they are also synchronized in frequency. For some applications frequency synchronization may be adequate; for others a combination of frequency and time/phase may be required. For some applications, the source of time/timing may be specified, and for others the source could be any one of a set of (master) clocks.

The three use case categories considered here are:

- 1) Wireless
- 2) Infrastructure
- 3) Media Gateway

IV.2 Wireless

IV.2.1 Applications

Within this general use case category, there are several applications of importance. Some of these require just frequency information, and others require time-of-day, and others require phase. The application, from the viewpoint of timing, is to deliver the appropriate timing information to a base station (e.g., Node B).

IV.2.2 Examples

IV.2.2.1 GSM base station (frequency synchronization)

The timing requirement applicable to the GSM radio interface can be found in [b-ETSI TS 145010]. The radio interface requirement for a GSM base station is frequency accuracy of ± 50 ppb. In case of Pico base stations, the accuracy can be relaxed to ± 100 ppb. The need for this requirement stems primarily from the need to support handover of mobiles between base stations. It should be noted that relevant requirements documents do not directly address the (wireline) network interface. Nevertheless in case of TDM networks, the synchronization requirements on input signals are normally expressed in terms of output wander masks presented in [ITU-T G.823] and [ITU-T G.824], and traceability to a PRC source.

It should be noted that in case of GSM radio access network, there are not so strict frequency accuracy requirements related to limit the slip rate. In fact, in these cases, the data of a single user is stored in relatively large buffer (from 10 to 30 ms) and also assuming 50 ppb frequency accuracy the data would be lost (buffer empty or full) after long times, much longer if compared with classical switching network elements where buffers handling the data are much smaller (125 μ s).

IV.2.2.2 UMTS FDD base station (frequency synchronization)

The timing requirement applicable to the WCDMA FDD radio interface can be found in [b-ETSI TS 125104].

The radio interface requirement for UMTS FDD base stations is a frequency accuracy of ± 50 ppb; for the FDD mode, there are no phase alignment requirements.

As for the case of GSM networks, there are not so strict frequency accuracy requirements related to limit the slip rate because of the large buffer used to store data of a single user.

IV.2.2.3 UMTS TDD base station (frequency and phase synchronization)

The timing requirement applicable to the WCDMA TDD radio interface can be found in [b-ETSI TS 125105].

The radio interface requirement for UMTS TDD base stations is a frequency accuracy of ± 50 ppb; for the TDD mode, there is the additional requirement for the phase alignment of neighbouring base stations to within 2.5 μ s.

As for the case of GSM networks, there are not so strict frequency accuracy requirements related to limit the slip rate because of the large buffer used to store data of a single user.

IV.2.2.4 3GPP2 CDMA2000 base station (frequency and time synchronization)

The relevant CDMA2000 standards are the [b-3GPP2 C.S0010-B] and [b-3GPP2 C.S0002-C].

According to the CDMA2000 specifications, the average frequency difference between the actual CDMA transmit carrier frequency and specified CDMA transmit frequency assignment shall be less than ± 50 ppb.

In the CDMA2000 specifications, it is also specified that each base station shall use a time base reference that is time-aligned to CDMA System Time. CDMA System Time is synchronous to UTC time (except for leap seconds) and uses the same time origin as GPS time. All base stations use the same System Time (within a small error tolerance). For all base stations, the pilot time alignment error should be less than 3 μ s and shall be less than 10 μ s.

Because of the above requirements, it is a common practice to equip CDMA base stations with GPS receivers.

IV.2.2.5 TD-SCDMA base station (frequency and phase synchronization)

The timing requirement applicable to the TD-SCDMA radio interface can be found in [b-3GPP TR 25.836].

The radio interface requirement for TD-SCDMA base stations is a frequency accuracy of ± 50 ppb; there is the additional requirement for the phase alignment of neighbouring base stations to within 3 μ s (this requirement is then measured comparing the phase between adjacent base stations).

Because of the above requirements, it is a common practice to equip TD-SCDMA base stations with GPS receivers.

IV.2.3 Remarks

The requirements listed in the previous clauses apply to the radio interface. When time or frequency reference is carried by the network, other requirements apply. These depend on several factors such as radio base station oscillator characteristics, filter capability of the radio base station, etc. As an example, a long-term frequency accuracy significantly better than 50 ppb may be required for the reference timing signal carried over the network in case of 50 ppb frequency accuracy requirement shall be fulfilled on the radio interface. The value of 16 ppb ([ITU-T G.812] Type II frequency accuracy) has sometimes been mentioned.

Similarly, in case accurate time and/or phase shall be distributed to the radio base stations, the budget to be allocated to the network might be much smaller than the requirements defined by the wireless standards to be fulfilled on the radio interface. These aspects are for further study.

In several cases, such as the GSM base station situations, such equipment is deployed and working and capable of deriving its timing needs from the traffic interface to the (wireline) network, such as PDH or SDH. If the PDH/SDH link is replaced by an Ethernet or synchronous Ethernet link, the needs of the base station shall still be met.

Distribution of phase/time is not common in the case of PDH/SDH links. Accurate phase and time is commonly distributed via GPS. Depending on the accuracy requirements and on the network conditions, methods based on time stamps (see clause 7.2) may be also appropriate for this purpose. In some implementations, two-way protocols are used.

IV.3 Infrastructure

There are several applications in this use case category including IP-DSLAM, Modular-CMTS, MSAN, OLT, etc. This area is for further study.

IV.4 Media gateway

For further study.

Appendix V

Packet networks reference models

(This appendix does not form an integral part of this Recommendation)

The packet network reference models that have been used to characterize the performance of the packet networks, in terms of packet delay variations, are shown in Figures V.1 and V.2: model A in Figure V.1 is related to applications with very strict delay and delay variation requirements, model B in Figure V.2 refers to scenarios with less strict packet delay variation requirements.

These models do not describe how packet networks have to be designed. The purpose of the models is purely to get a general understanding of the characteristics of typical packet networks.

V.1 Ethernet networks models

The following models have been defined for the case of Ethernet networks (Figures V.1 and V.2).

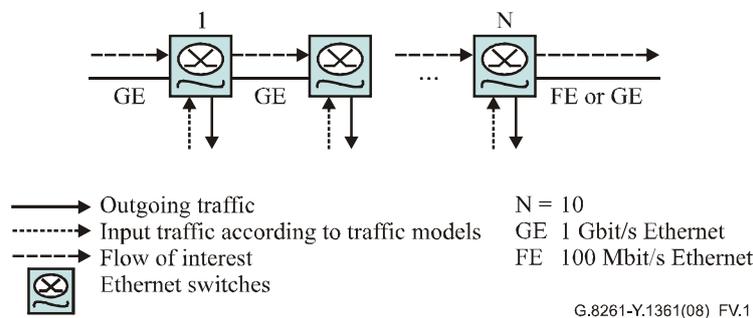


Figure V.1 – Packet network reference model A (switched Ethernet network)

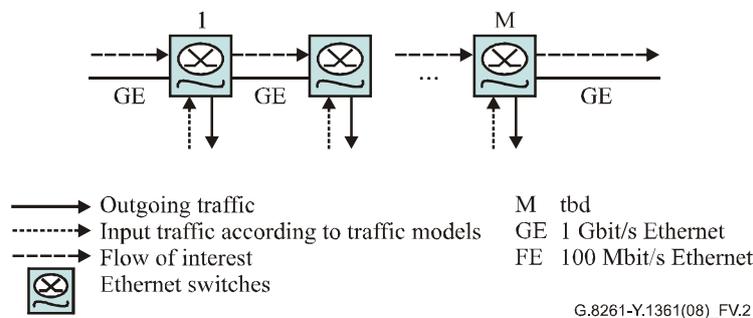


Figure V.2 – Packet network reference model B (switched Ethernet network)

NOTE 1 – With respect to the number of Ethernet switches ("M") in Figure V.2, there is a common agreement that 20 is a reasonable number. This has to be confirmed.

NOTE 2 – 10 Gbit/s links could be considered in new models.

The following cases have been considered:

- Scenario 1: Switched Ethernet network, Best effort with over-provisioning (single queue)
- Scenario 2: Switched Ethernet network, quality of service according to [IEEE 802.1Q], [b-IEEE 802.1p] (at least 2 queues, with one dedicated queue for handling real-time data and Weighted Fair Queuing (WFQ), discipline)
- Scenario 3: Switched Ethernet network, quality service according to [IEEE 802.1Q], [b-IEEE 802.1p] (with one queue dedicated for the handling of data used for timing recovery, e.g., Time stamps)

NOTE 3 – In order to understand the applicability of the models in Figures V.1 and V.2, a simple approach could be to define two main classes of network scenarios: a backbone network, that can also be used to carry services in the access network (e.g., leasing bandwidth), and a dedicated access network. Model B (Figure V.2), could be the reference model mainly applicable for the first type of packet network (backbone), while Model A (Figure V.1) could be the reference model mainly applicable to an access network (e.g., wireless access network).

Referring to the models described in clause 9, this means that, in general (most of the cases), the CE Island in Case 1 and Case 3 could be characterized by packet network reference Model B, while the CE Island in Case 2 could be characterized by packet network reference Model A. A third case is when bandwidth is leased by an operator to connect two end points connected via Ethernet switches (e.g., 100 Mbit/s guaranteed bandwidth over 1 Gbit/s transport). Also in this case, models in this appendix could be used. With a proper service level agreement between the customer and the Ethernet network operator, one could assume that the interfering traffic in the intermediate nodes could be considered as traffic with lower priority. The SLA in this case could guarantee bandwidth and increase priority, since both will be key elements of a premium SLA, such as, for instance, the cellular operators will require from their Ethernet providers. This could then be considered as a scenario with traffic handling characteristics between scenario 2 and scenario 3. With respect to the expected result, when leasing bandwidth in a packet network, better performance could then normally be achieved if compared with scenarios 1 and 2.

These are the conditions considered as a basis for the characterization of a packet network:

- Traffic load: 60% static
- Packet Rate: 10 packets per second
- Observation intervals: 60 minutes
- Traffic models according to Appendix VI
- Packet length 90 octets.

With respect to the conditions listed above, the characteristics of 2 Mbit/s signals may also be considered, i.e., packets with a payload of 256 octets and packet rate of 1000 p/s.

Based on the above models, the following parameters describe the typical behaviour of the packet network in the different cases:

Table V.1 – Parameters for the relevant network models

Network model		Average delay (μ s)	min delay + Threshold (Note) (x%) (μ s)
Model A	Scenario 1	1400	800 + 1700 (95%) 800 + 800 (50%) 800 + 20 (10%) 800 + 1 (1%)
	Scenario 2	ffs	Ffs
	Scenario 3	ffs	Ffs
Model B	Scenario 1	ffs	Ffs
	Scenario 2	ffs	Ffs
	Scenario 3	ffs	Ffs
NOTE – This value is the maximum delay variation for x% of the packets (95%, 50%, 10% and 1% are the reference values).			

NOTE 4 – The values are based on a configuration with only 100 Mbit/s links. This provides a conservative scenario, especially for packets with higher delay variation. Further work is needed in order to confirm and complete the table.

Details on the test cases that are needed to test the network also in non-static or failure conditions are provided in Appendix VI.

Different packet rates may be used in order to test different applications and to improve the performance of the filtering algorithms (this is relevant for adaptive methods, or more, in general, when the synchronization is carried over packets).

Spectral distribution shall also be considered in the characterization of the packet networks. For this purpose, some initial metrics have been proposed and are under study.

Among these, the minTDEV operator has been defined based on the TDEV metric. The TDEV metric is shown below in Equation V-1:

$$\sigma_x(\tau) = TDEV(\tau) = \sqrt{\frac{1}{6} \left\langle \left[\frac{1}{n} \sum_{i=1}^n x_{i+2n} - 2 \frac{1}{n} \sum_{i=1}^n x_{i+n} + \frac{1}{n} \sum_{i=1}^n x_i \right]^2 \right\rangle} \quad (V-1)$$

The TDEV operator is based on the mean of the sample window (Equation V-2):

$$x_{mean}(i) = \frac{1}{n} \sum_{j=1}^n x_{j+i} \quad (V-2)$$

Compared with the TDEV operator, in the minTDEV operation the mean of the sample window is replaced with the minimum of the sample window as shown in Equation V-3:

$$x_{min}(i) = \min[x_j] \text{ for } (i \leq j \leq i+n) \quad (V-3)$$

Substituting Equation V-3 back into original TDEV definition yields:

$$\sigma_{x_min}(\tau) = \min TDEV(\tau) = \sqrt{\frac{1}{6} \left\langle [x_{min}(i+2n) - 2x_{min}(i+n) + x_{min}(i)]^2 \right\rangle} \quad (V-4)$$

Equation V-4 defines the minTDEV operator.

The minTDEV operator has been indicated as a useful tool in combination with packet networks that exhibit a PDV behaviour, where it is possible to identify a suitable set of packets with packet delay variation close to a minimum delay.

In fact, these packets are less impacted by the queuing delays, and therefore are more representative of the original timing.

Because of its definition, the minTDEV may not fully address all network scenarios (e.g., those with two-sided PDV distributions) and further study is needed. In addition, it shall be noted that minTDEV shall be used in combination with other complementary metrics such as TDEV.

The minTDEV operator has also been proposed to validate the test set-up described in Appendix VI using some minTDEV mask. This area is also for further study.

Since TDEV and minTDEV are defined to operate on evenly spaced samples and packet-based timing flows are not constrained to be evenly spaced, there is a need to define the interpolation approach for general application. In many practical cases the timing flow periodicity is sufficient such that no interpolation is required. This is an area for further study.

Packet timing flows through networks experience delay variation but no long-term frequency offset or frequency drift. However, there may be offset or drift between the clock at the source, as compared to the clock in the measurement device. Both TDEV and minTDEV are sensitive to drift

while minTDEV is also sensitive to frequency offset. Depending on the severity of these effects, techniques need to be defined to mitigate these effects. This is also an area for further study.

NOTE 5 – The definition of other scenarios is under study. These models should allow the study of other meaningful network scenarios.

V.2 Other network models

Other network models can be defined based on the considerations provided by this clause.

In particular, this clause emphasizes on the compound networks that may support circuit emulation services, showing that the various network designs may introduce new variables to timing transmission, performance and test scenarios.

NOTE 1 – The TDM PW (TDM pseudowire) terminology is used in other contexts to describe the transmission of TDM over packet network, and will be used in this clause as a different way to address the CES aspects.

In particular, the network scenarios presented here show that:

- TDM PW may go over a unique domain made of a unique transport technique (Ethernet, IP or MPLS)
- TDM PW may go over a unique domain made of diverse transport techniques
- TDM PW may go over different domains made of unique or diverse transport techniques
- TDM PW crossing different domains or transport techniques may imply modifying the IWF packet layers (e.g., IP to MPLS).

For TDM PW timing using adaptive clock recovery model, the diversity of equipment, policy (e.g., QoS) and transmission methods may impact the recovered timing quality.

The examples given in this clause are the most common expected to be deployed. However, these do not intend to cover any possible scenarios, such as when using traffic engineering tunnelling (stacking MPLS labels or [b-IEEE 802.1ah]) or layering (GFP, T-MPLS).

Deployed networks are made of different technologies. Considering a TDM PW for instance, a TDM service set-up between two IWFs may go over multiple transmission technologies and network domain.

Some examples are provided below.

At access, it may be an Ethernet network made of Ethernet switches only, as described in Figure V.3.

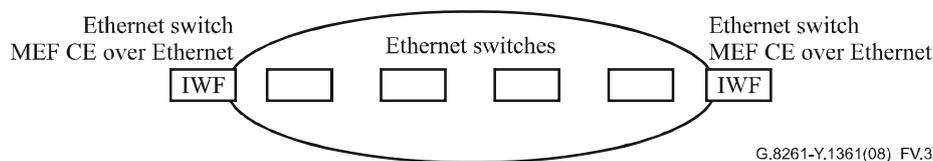


Figure V.3 – Ethernet switches only network

NOTE 2 – The scenario shown in this figure can be modelled with the reference models shown in Figures V.1 and V.2.

It could also be an MPLS network with P devices and IWF at PE (provider edge), as shown in Figure V.4.

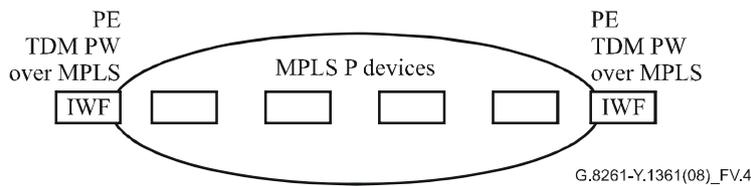


Figure V.4 – MPLS PE/P only network

It could also be an IP network with IP routers and IWF in routers as shown in Figure V.5.

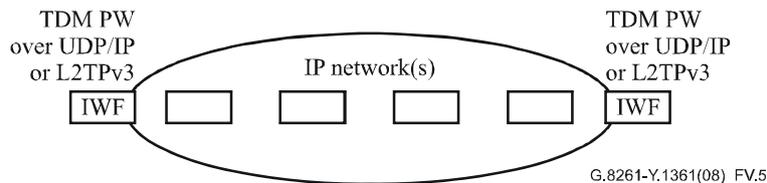


Figure V.5 – IP routers only network

NOTE 3 – The network characteristics in terms of packet delay variation of the scenarios shown in Figures V.4 and V.5 (except when a software-based forwarder is used) can be based on the results from the models shown in Figures V.1 and V.2.

However, current networks are often more complex. Within one domain or operator, it may consist of different transport technologies. A TDM PW may also cross different domains.

Some examples are given below.

- 1) TDM MPLS PW crossing 3rd party MPLS carrier (Figure V.6)

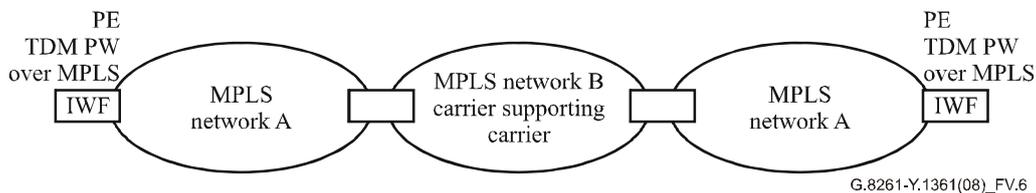


Figure V.6 – MPLS network over MPLS network

- 2) TDM MPLS PW terminated on distinct carrier IWF devices (Figure V.7)

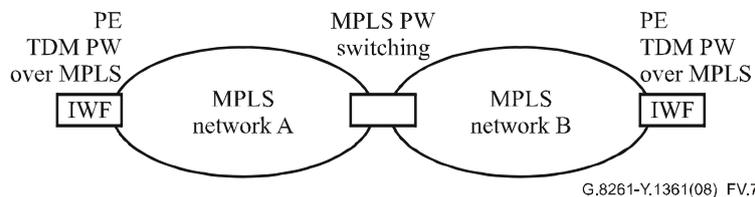


Figure V.7 – Crossing distinct MPLS networks or domains

NOTE 4 – Such scenario may also depict a change in transport layer as shown in Figure V.8, where the TDM PW moves from MPLS to IP. In this case, the TDM packet encapsulation payload is not changed. Only the PSN layer changes.

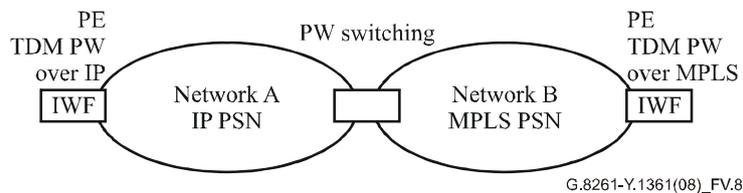


Figure V.8 – Swapping the PSN layers

NOTE 5 – it may be possible that the TDM stream may have to be recovered at an interconnect point between two domains or operators, either because the previous scenario is not possible (different TDM PW encapsulations), or because operators do not agree on the interconnection method (can be with respect to location or management of the switching node, the encapsulation or control plane). This is shown in Figure V.9.

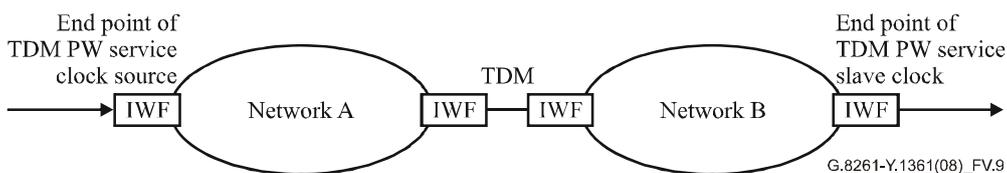


Figure V.9 – Crossing distinct operator networks with no PW switching function

- 3) TDM IP PW using MPLS network, optionally using a L3VPN service (Figure V.10)

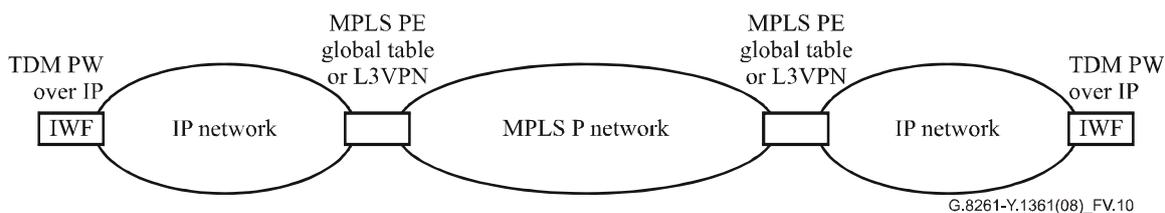


Figure V.10 – IP over MPLS network

- 4) TDM Ethernet PW using MPLS network for transmission (Figure V.11)

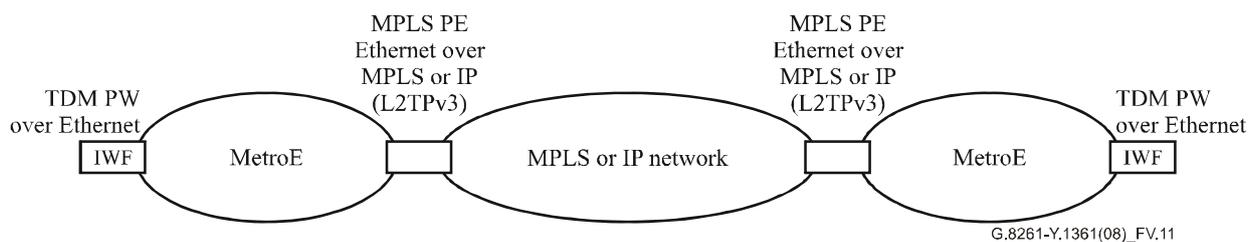


Figure V.11 – Ethernet over MPLS or IP network

- 5) TDM IP PW using Ethernet PW service over MPLS network (Figure V.12)

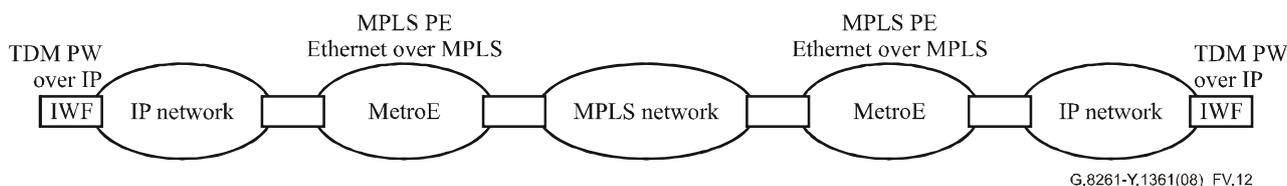


Figure V.12 – IP over Ethernet over MPLS network

Some key aspects of such compound networks include:

- Network equipments will have different characteristics;
- Network policy (e.g., QoS) may be different if crossing different domains;
- Timing architecture may be different.

Appendix VI

Measurement guidelines for packet-based methods

(This appendix does not form an integral part of this Recommendation)

The guidelines in this appendix are intended to assist in the acquisition of performance characteristics used to establish benchmarking results.

It is important to consider that, when doing performance comparisons, the configurations of the systems being compared be as similar as possible.

Results from the test cases provided in this appendix provide no guarantee that equipment will perform as expected in a complex network situation under a range of complex and changing load conditions.

Although test cases in this appendix provide a useful guidance on the performance of Ethernet-based circuit emulation techniques, evaluation in complex network scenarios that mimic the deployment profile is strongly recommended.

VI.1 Measurement reference points

The measurement reference points are provided in Figure VI.1 (differential clock recovery method) and Figure VI.2 (adaptive clock recovery method). The two figures of this clause provide two of the most relevant test scenarios. Additional scenarios may be identified in future versions of this Recommendation.

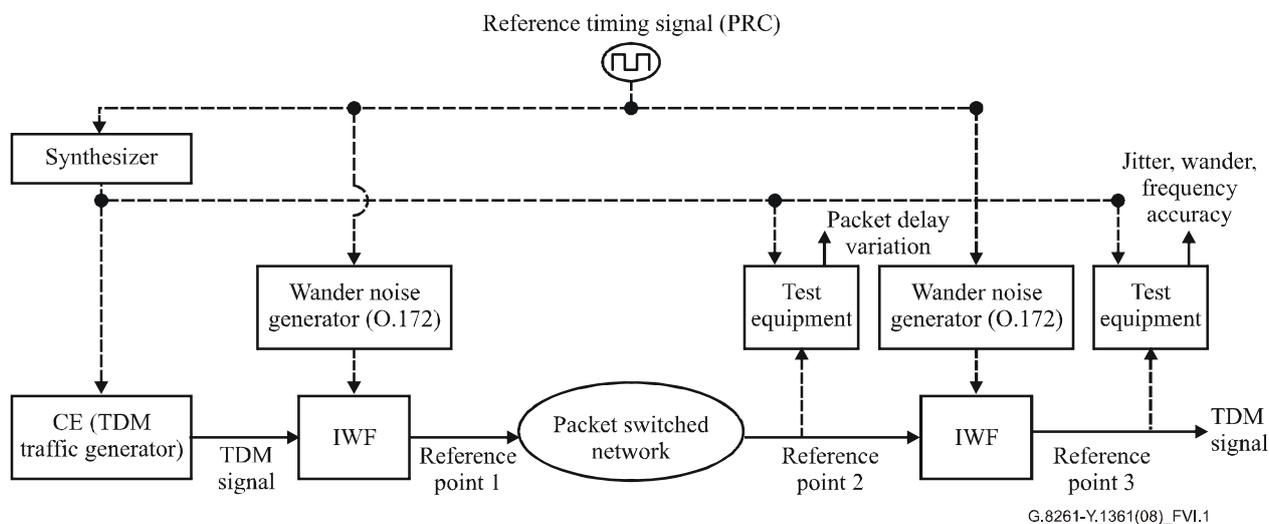
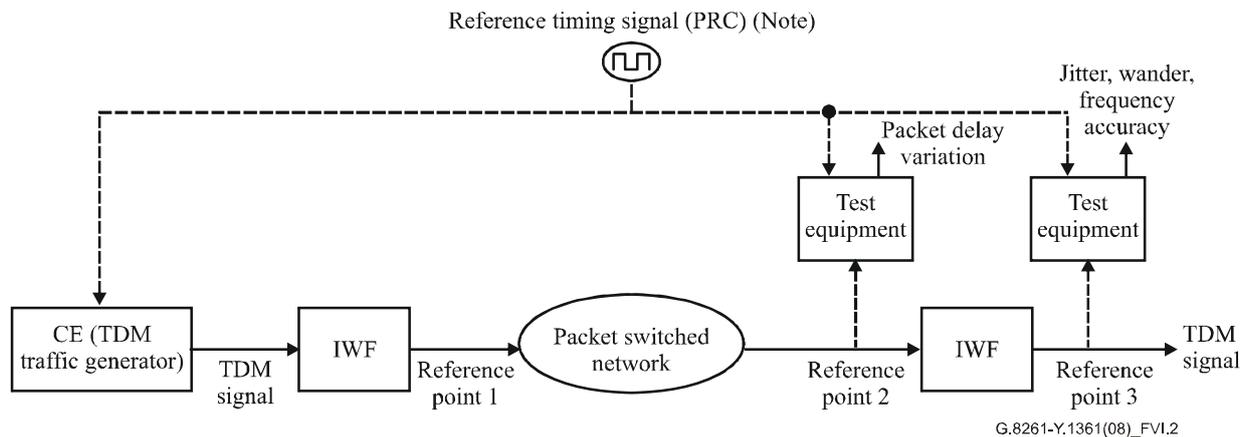


Figure VI.1 – Measurement reference points in the differential clock recovery method



NOTE – The reference timing signal (PRC) is used to represent the TDM service clock.

Figure VI.2 – Measurement reference points in the adaptive clock recovery method

NOTE 1 – the "Wander Noise Generator" in Figure VI.1 is inserted to simulate the noise generated by the synchronization network (as specified in [ITU-T O.172]). The output of the wander noise generation should comply with the synchronization interface as specified in [ITU-T G.824] and [ITU-T G.823].

NOTE 2 – The synthesizer in Figure VI.1 is needed to change the frequency of the asynchronous TDM signals (within the G.703 limits).

NOTE 3 – This appendix contains a suite of tests to evaluate the performance of clock recovery under different kinds of network topologies, traffic characteristics and impairments. However, the tests defined here are not exhaustive, and do not cover all possible impairments that may be caused by the packet network. Further tests may be defined in future, for example:

- Clock recovery under the presence of link aggregation, such as [IEEE 802.1ad]
- Clock recovery under the presence of QoS
- Clock recovery under the presence of flow control, such as IEEE 802.3x pause frames

NOTE 4 – Measurement methodologies for the asynchronous signals are provided in Appendix II of [ITU-T G.823].

VI.2 Input traffic characteristics

To be able to account for different traffic types in the network, two types of disturbance traffic models are defined as described in clauses VI.2.1 and VI.2.2 below.

The Network Traffic Model 1 is intended to model the traffic in the access network where the majority of the traffic is voice. The Network Traffic Model 2 is intended to model the traffic on networks where the majority of the traffic is data.

It should be noted that the CES traffic is in addition to the disturbance traffic.

NOTE 1 – The details on how traffic is injected have to be provided when performing the tests. The details should cover aspects such as how the traffic is mixed, which Ethernet switches are receiving the traffic, the packet rate for the CBR flows, etc. As an example for details on how the traffic is mixed, the following approach could be followed:

- the different packet size profiles will appear in a random manner with probability 0.8, 0.15 and 0.05 respectively. The random generation process will be identically independent distributed (uncorrelated) based on some PRBS with a minimal period of 223-1 frames.

Maximum size packets will occur in bursts lasting between 0.1 s and 3 s. For each burst event, the burst length will be selected randomly using an identically independent uniformly distributed random generator between 0.1 s to 3 s.

NOTE 2 – The traffic can be injected in serial (into one port of the Ethernet switch) or in parallel (into multiple ports of the Ethernet switch) and, in general, different behaviour is expected. However, similar statistical properties of the packet delay variation at the output of the packet network have been observed for

the serial and parallel cases when the statistics of the packets with minimum delay were not significantly impacted by the load conditions. The following are some of the aspects that may impact the statistics of the packets with minimum delay:

- queuing strategy in the switches
- number of switches in chain
- static vs non-static load

VI.2.1 Network Traffic Model 1

According to 3GPP, the access traffic is composed by conversational (voice), streaming (audio-video), interactive (e.g., http) and background (sms, e-mail). It is known that in wireless network 80% to 90% of the traffic is conversational, with the average call lasting from 1 minute to 2 minutes. To be able to model this traffic, 80% of the load should be based on packets of fixed small size constant bit rate, and 20% based on packets with a mix of medium and maximum size.

The packet size profile is:

- 80% of the load must be minimum size packets (64 octets)
- 15% of the load must be maximum size packets (1518 octets)
- 5% of the load must be medium size packets (576 octets)

Maximum size packets will occur in bursts lasting between 0.1 s and 3 s.

VI.2.2 Network Traffic Model 2

Bigger packets compared with the Network Traffic Model 1 compose the network that handles more data traffic. To be able to model this traffic, 60% of the load should be based on packets of maximum size, and 40% on packets with a mix of minimum and medium size.

The packet size profile is:

- 60% of the load must be maximum size packets (1518 octets)
- 30% of the load must be minimum size packets (64 octets)
- 10% of the load must be medium size packets (576 octets)

Maximum size packets will occur in bursts lasting between 0.1 s and 3 s.

NOTE – Traffic Model 1 is based on the typical traffic characteristics of the wireless access networks. However, there are cases when, in order to optimize the bandwidth usage during the traffic peak hours, the packets to/from base stations with Ethernet interface can be bundled in packets of larger size resulting in traffic characteristics pertaining to those of Traffic Model 2 instead. In this case, the traffic models characteristics may change over time.

VI.3 Test topologies for adaptive methods

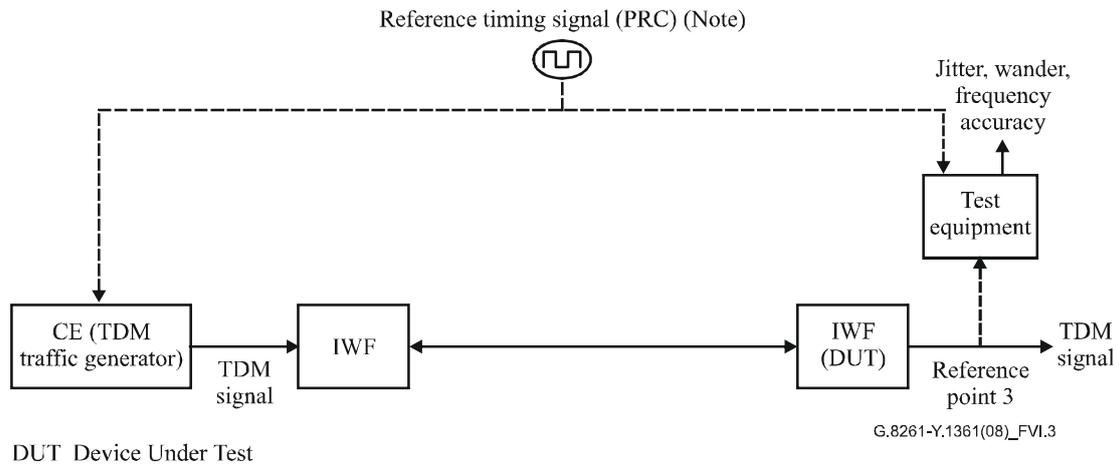
The testing topologies described herein include methods for testing the synchronization methods applicable to this Recommendation.

These tests have been defined in a controlled environment (i.e., not in the field).

NOTE – The test cases presented in this clause address the test of the CES domain. The test of the PNT domain when adaptive clock recovery methods are used could be done using the same approach. Some adaptation of the test cases set-up might be needed for this purpose. This is for further study.

VI.3.1 Baseline test

Baseline test topology is shown in Figure VI.3.



NOTE – The reference timing signal (PRC) is used to represent the TDM service clock.

Figure VI.3 – Baseline test topology

The baseline test should be done under the following conditions:

- No packet load
- Test measurements
 - Measure TIE, MTIE, and MRTIE (as described in [ITU-T G.823] and [ITU-T G.824])
 - Measure frequency accuracy (the value for the frequency accuracy measurement integration-time is dependent upon the relevant end equipment)
 - Performance should meet the network limits for the relevant cases as defined in clause 9.

VI.3.2 Performance test

The performance test is equivalent to Model A in Appendix V, composed with either 10 gigabit Ethernet switches or 9 gigabit Ethernet switches and 1 fast Ethernet switches. The test topology is shown in Figure VI.4.

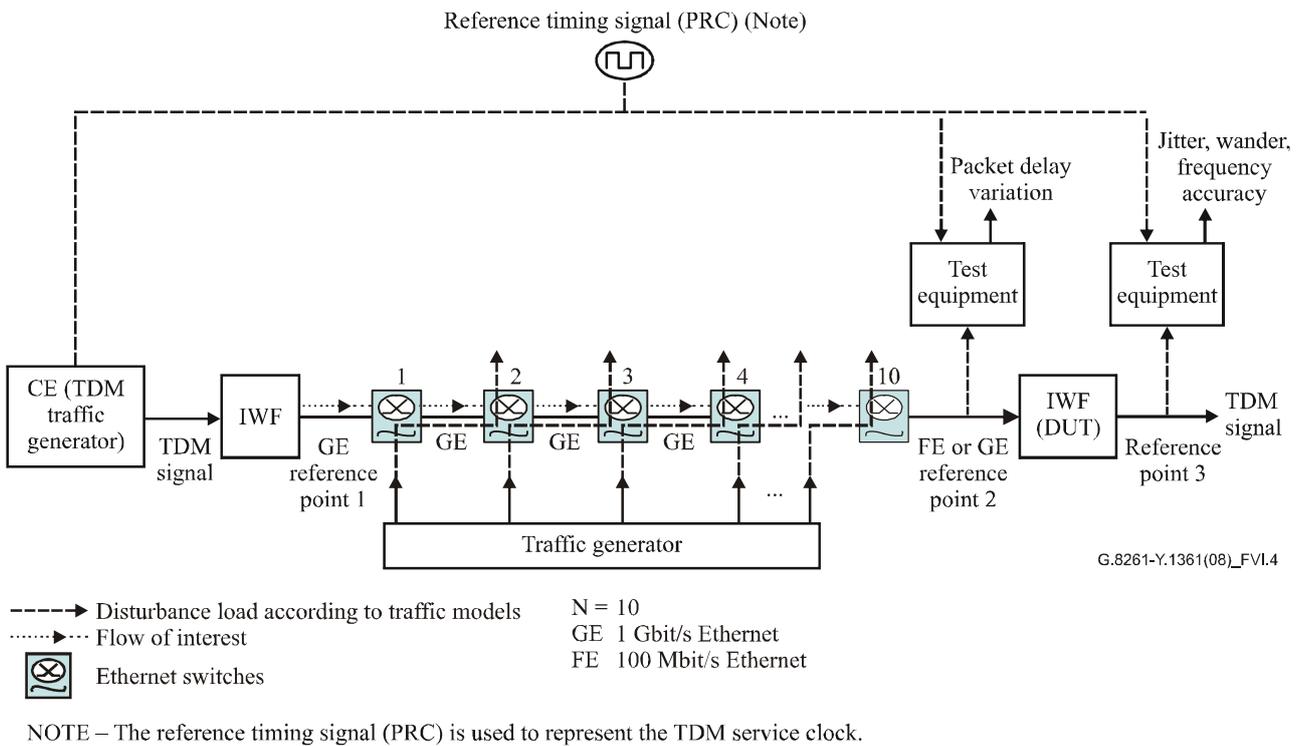


Figure VI.4 – Performance test topology

A specific test topology shown in Figure VI.5 is needed to perform the test case on traffic concentration forming bottleneck. This kind of configuration creates the beating effect (see Figures 20 and 21).

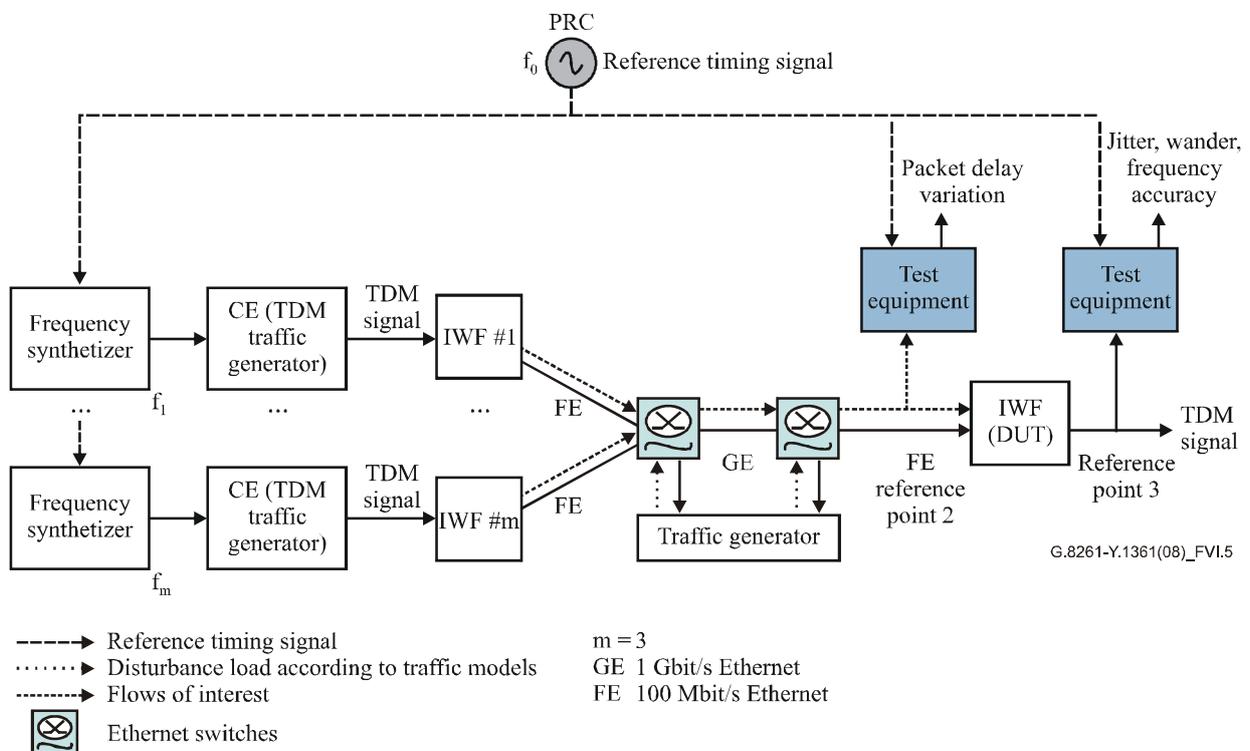


Figure VI.5 – Performance test topology for traffic concentration test case

The DUT must be tested for stability of operation under disruptive events that may cause the synchronization to fail or go out of specification. Test cases described in this clause are performed to test the DUT under load variation, network changes and packet loss.

For each of the test cases described in this clause, the following measurements should be performed:

- Measure TIE, MTIE, and MRTIE (as described in [ITU-T G.823] and [ITU-T G.824])
- Measure frequency accuracy (the value for the frequency accuracy measurement integration-time is dependent upon the relevant end equipment)
- Measure packet delay variation
- Performance should meet the network limits for the relevant cases as defined in clause 9.

NOTE 1 – The test set-up, as described in Figure VI.4, provides the starting point towards a common test scenario.

However, in order to get a test environment that will be simpler to be implemented and in order to remove any risk for getting different results when using Ethernet switches of different technologies, a proposal is under discussion to replace the specification as defined in Figure VI.4 with a new test set-up, where in place of the Ethernet switches and the traffic generator, the delay variation could be created by a test equipment with a delay variation profile as input.

This delay variation profile could be expressed in terms of delay variation "test vectors" (test sequence) of duration 15 min, 60 min, and 24 hours. The delay variation shall be expressed with the proper timing resolution.

The test sequences would be based on the results from the tests performed using the tests topology as described in Figure VI.4.

NOTE 2 – Deterministic test cases may also be considered in addition to the test cases described in this clause. This is for further study.

VI.3.2.1 Test Case 1

Test Case 1 models the "Static" Packet load. Test Case 1 must use the following network conditions:

- Network disturbance load with 80% for 1 hour. The test measurements should start after the clock recovery is in a stable condition. Guidance on stabilization period is provided in Appendix II. The disturbance background traffic to load the network must use network Traffic Model 2 as defined in clause VI.2.2.

VI.3.2.2 Test Case 2

Test Case 2 models sudden large and persistent changes in network load. It demonstrates stability on sudden large changes in network conditions, and wander performance in the presence of low frequency PDV.

Test Case 2 must use the following network conditions:

- The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
- Allow a stabilization period according to Appendix II for the clock recovery process to stabilize before doing the measurements.
- Start with network disturbance load at 80% for 1 hour, drop to 20% for an hour, increase back to 80% for an hour, drop back to 20% for an hour, increase back to 80% for an hour, drop back to 20% for an hour (see Figure VI.6).

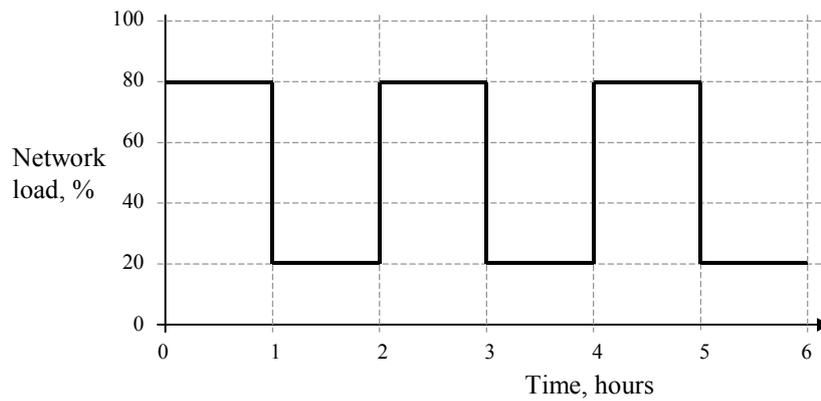


Figure VI.6 – Sudden network disturbance load modulation

- Repeat the test using the Network Traffic Model 2 as defined in clause VI.2.2 to load the network.

VI.3.2.3 Test Case 3

Test Case 3 models the slow change in network load over an extremely long timescale. It demonstrates stability with very slow changes in network conditions, and wander performance in the presence of extremely low frequency PDV.

Test Case 3 must use the following network conditions:

- The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
- Allow a stabilization period according to Appendix II for the clock recovery process to stabilize before doing the measurements.
- Vary network disturbance load smoothly from 20% to 80% and back over a 24-hour period (see Figure VI.7).

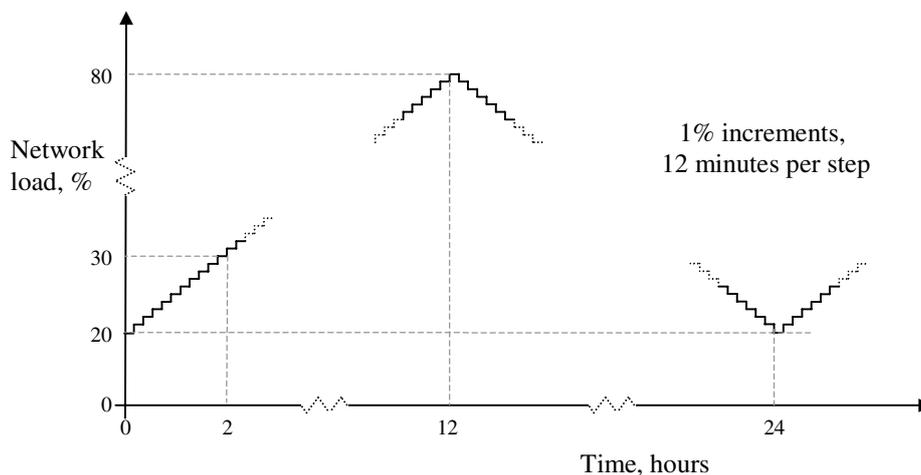


Figure VI.7 – Slow network load modulation

- Repeat the test using the Network Traffic Model 2 as defined in clause VI.2.2 to load the network.

VI.3.2.4 Test Case 4

Test Case 4 models temporary network outages and restoration for varying amounts of time. It demonstrates ability to survive network outages and recover on restoration. It should be noted that

MTIE over the 1000 s interruption will largely be governed by the quality of the local oscillator, and should not be taken as indicative of the quality of the clock recovery process.

Test Case 4 must use the following network conditions:

- The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
- Start with 40% of network disturbance load. After a stabilization period according to Appendix II, remove network connection for 10 s, then restore. Allow a stabilization period according to Appendix II for the clock recovery process to stabilize. Repeat with network interruptions of 100 s.
- Repeat the test using the Network Traffic Model 2 as defined in clause VI.2.2 to load the network.

VI.3.2.5 Test Case 5

Test Case 5 models temporary network congestion and restoration for varying amounts of time. It demonstrates the ability to survive temporary congestion in the packet network.

Test Case 5 must use the following network conditions:

- The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
- Start with 40% of network disturbance load. After a stabilization period according to Appendix II, increase network disturbance load to 100%, (inducing severe delays and packet loss) for 10 s, then restore. Allow a stabilization period according to Appendix II for the clock recovery process to stabilize. Repeat with a congestion period of 100 s.
- Repeat the test using the Network Traffic Model 2 as defined in clause VI.2.2 to load the network.

VI.3.2.6 Test Case 6

Test Case 6 models routing changes caused by failures in the network.

Test Case 6 must use the following network conditions:

- Change the number of switches between the DUTs, causing a step change in packet network delay.
 - The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
 - Start with 40% of network disturbance load. After a stabilization period according to Appendix II, re-route the traffic to bypass one switch in the traffic path. This shall be done by updating the test set-up in Figure VI.4 adding a cable from switch in position "n" to switch in position "n+2", and either using a fibre spool or adding an impairment box able to simulate different cable lengths (10 μ s and 200 μ s can be simulated as typical examples). The configuration shall be done so that the traffic flow under test is routed directly from switch in position "n" via the new link to switch in position "n+2".
After disconnecting the cable from switch "n" to switch "n+2" (so that traffic under test will then be routed from switch in position "n" to switch in position "n+1"), allow a stabilization period according to Appendix II for the clock recovery process to stabilize, and then reconnect the link that was disconnected in order to restore the traffic on the original path.
 - Start with 40% of network disturbance load. After a stabilization period according to Appendix II, re-route the traffic to bypass three switches in the traffic path. This shall be done by updating the test set-up in Figure VI.4 adding a cable from switch in position "n" to switch in position "n+4", and either using a fibre spool or adding an impairment

box able to simulate different cable lengths (10 μ s and 200 μ s can be simulated as typical examples). The configuration shall be done so that the traffic flow under test is routed directly from switch in position "n" via the new link to switch in position "n+4".

After disconnecting the cable from switch "n" to switch "n+4" (so that traffic under test will then be routed from switch in position "n" to switch in position "n+1"), allow a stabilization period according to Appendix II for the clock recovery process to stabilize, and then reconnect the link that was disconnected in order to restore the traffic on the original path.

- Repeat the test using the Network Traffic Model 2 as defined in clause VI.2.2 to load the network.

VI.3.2.7 Test Case 7

Test Case 7 models the beating effect due to traffic concentration with TDM sources of different frequency. In particular, this test case is referring to CES TDM flows associated with a 2048 Mbit/s or 1544 Mbit/s bitstream. The test set-up is shown in Figure VI.5 and must use the following network conditions:

- Network disturbance load with 60% for the entire test period. The test measurements should start after the clock recovery is in a stable condition and should run over 24 hours. Guidance on stabilization period is provided in Appendix II. The disturbance background traffic to load the network must use Network Traffic Model 2 as defined in clause VI.2.2.
- Apply the following frequencies with the frequency synthesizers to test the case of asynchronous services:

$$f_1 = f_0$$

$$f_2 = f_0 + 1 \text{ ppm}$$

$$f_3 = f_0 - 50 \text{ ppm (2048 kbit/s signals) or } f_0 - 32 \text{ ppm (1544 kbit/s signals)}$$

- At the output of the IWF on the right (Reference Point 3), select the TDM output signal sent by IWF #0 for measurement of the applicable jitter and wander limits and the TDM output signal sent by IWF #3 for measurement of the asynchronous service clock.
- Run the test again with the following frequencies to test the case of clocks in holdover mixed with asynchronous services:

$$f_1 = f_0$$

$$f_2 = f_0 + 16 \text{ ppb}$$

$$f_3 = f_0 - 50 \text{ ppm (2048 kbit/s signals) or } f_0 - 32 \text{ ppm (1544 kbit/s signals)}$$

NOTE 1 – Packet size must be the same for all CES packet streams.

NOTE 2 – The same test scenario can be also used to test different CES TDM bitstreams (e.g., DS3 CES).

NOTE 3 – Other test cases based on this test case (e.g., non-static tests where the frequency offset drifts over time) are for further study.

VI.3.2.8 Test Case 8

Test Case 8 models a topology-dependent mechanism in packet networks that can delay packets by more than would be expected from volume-of-traffic considerations alone (see clause 10.1.2.6).

The test network is as given in clause VI.3.2, Figure VI.4, with the following change: there is just one source of "disturbance traffic", which is injected at switch 1, and traverses the entire network exiting at switch 10 on a separate port to the time-sensitive traffic.

- Test Case part A
This test case is similar to Test Case 3 (clause VI.3.2.3). It tests for gradual increases and decreases in traffic load in the presence of the blocking effect as described in

clause 10.1.2.6. It is not thought necessary to go to the same extremely low frequency, and hence long test runs as Test Case 3 in order to demonstrate resilience to this particular effect. Using network traffic model 2, start with 0% disturbance traffic load. Allow an initial stabilization period, according to Appendix II. Then increase the traffic load in 1% increments every minute up to 50% load. Reduce the load again in 1% increments back down to 0%.

- Test Case part B

This test case is similar to Test Case 2 (clause VI.3.2.2). It tests for sudden increases and decreases in traffic load in the presence of the blocking effect as described in clause 10.1.2.6.

Using network traffic model 2, start with 0% disturbance traffic load. Allow an initial stabilization period, according to Appendix II. Then increase the traffic load up to 50% for one hour. Repeat three times.

VI.4 Test Topologies for differential methods

The test topology is shown in Figure VI.8.

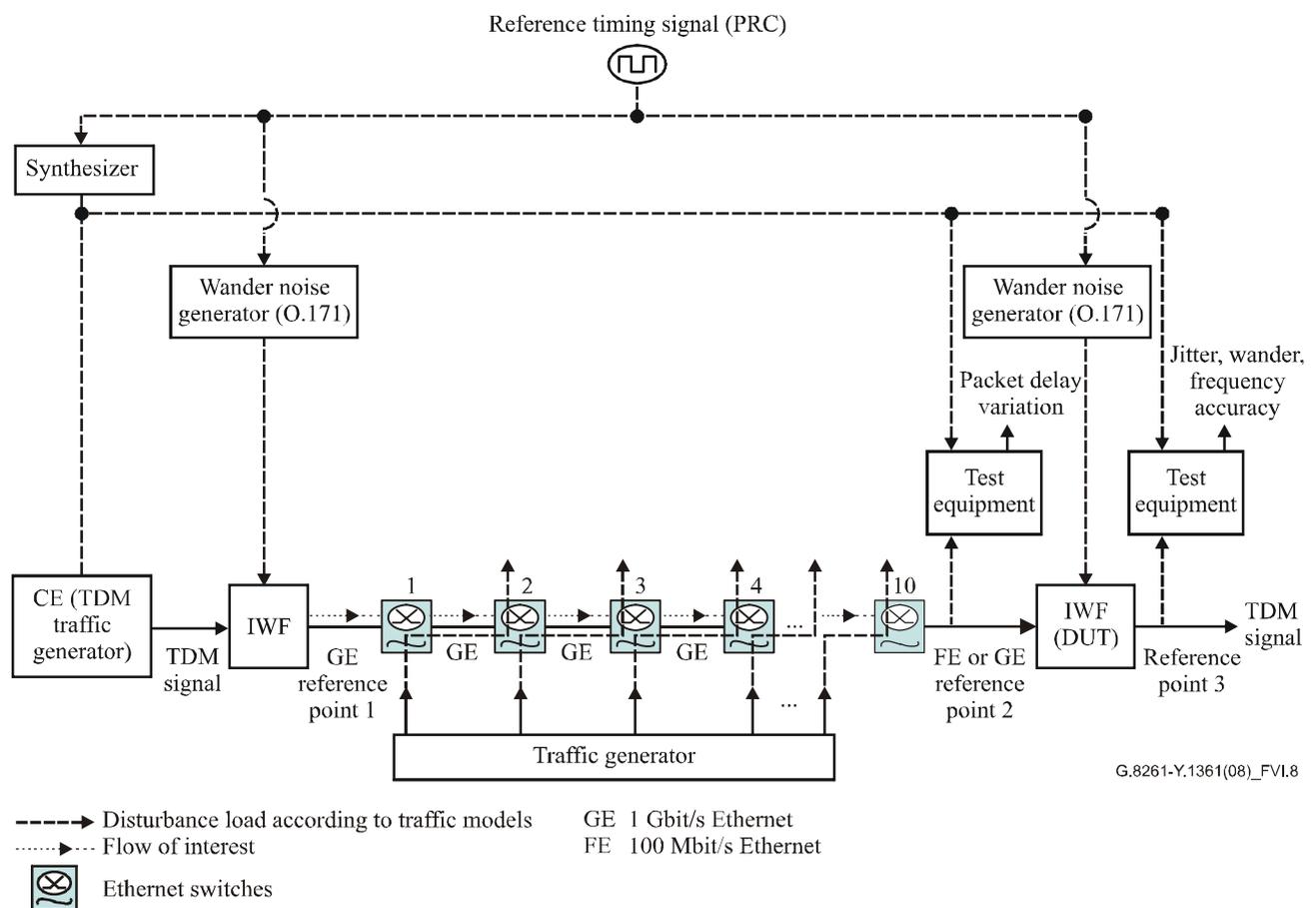


Figure VI.8 – Performance test topology for differential clock recovery method

NOTE – The frequency offset from the PRC introduced by the synthesizer should be + (or –) 50 ppm (2048 kbit/s) and + (or –) 32 ppm (1544 kbit/s) for all the test cases.

VI.4.1 Test Case 9

Test Case 9 models the performance of the differential clock recovery method under the "Static" Packet load. Test Case 9 must use the following network conditions:

- Network disturbance load with 80% for 1 hour. The test measurements should start after the clock recovery is in a stable condition. Guidance on stabilization period is provided in Appendix II. The disturbance background traffic to load the network must use Network Traffic Model 2 as defined in clause VI.2.2.

VI.4.2 Test Case 10

Test Case 10 models the performance of the differential clock recovery method with noise added into the reference timing signal into the IWF. It is used to simulate the noise generated by the Synchronization Network (as specified in [ITU-T O.172]).

Test Case 10 must use the following network conditions:

- Inject wander noise according to Annex C of [ITU-T O.172] to simulate the wander noise generated by the synchronization network. The actual values for the wander noise depends on the application (e.g., E1, DS1). The applicable wander noise masks are for further study.
- Network disturbance load with 80% for 1 hour assuming that the clock recovery is in a stable condition. Allow a stabilization period according to Appendix II for the clock recovery process to stabilize before doing the measurements. The packets to load the network must use Network Traffic Model 2 as defined in clause VI.2.2.

VI.4.3 Test Case 11

Test Case 11 models the performance of the differential clock recovery method with temporary network congestion and restoration for varying amounts of time. It demonstrates the ability to survive temporary congestion in the packet network.

Test Case 11 must use the following network conditions:

- The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
- Start with 40% of network disturbance load. After a stabilization period according to Appendix II, increase network disturbance load to 100% (inducing severe delays and packet loss) for 10 s, then restore. Allow a stabilization period according to Appendix II for the clock recovery process to stabilize. Repeat with a congestion period of 100 s.

NOTE – For the differential method, the following test cases have also been identified as relevant: Holdover (reference timing signal loss); different QoS cases. These are for further study.

VI.5 Test for two-way protocols

The testing topologies described herein include methods for testing two-way synchronization methods (e.g., time distribution protocols) applicable to this Recommendation.

These tests have been defined in a controlled environment (i.e., not in the field).

VI.5.1 Baseline test

Baseline test topology is shown in Figure VI.9.

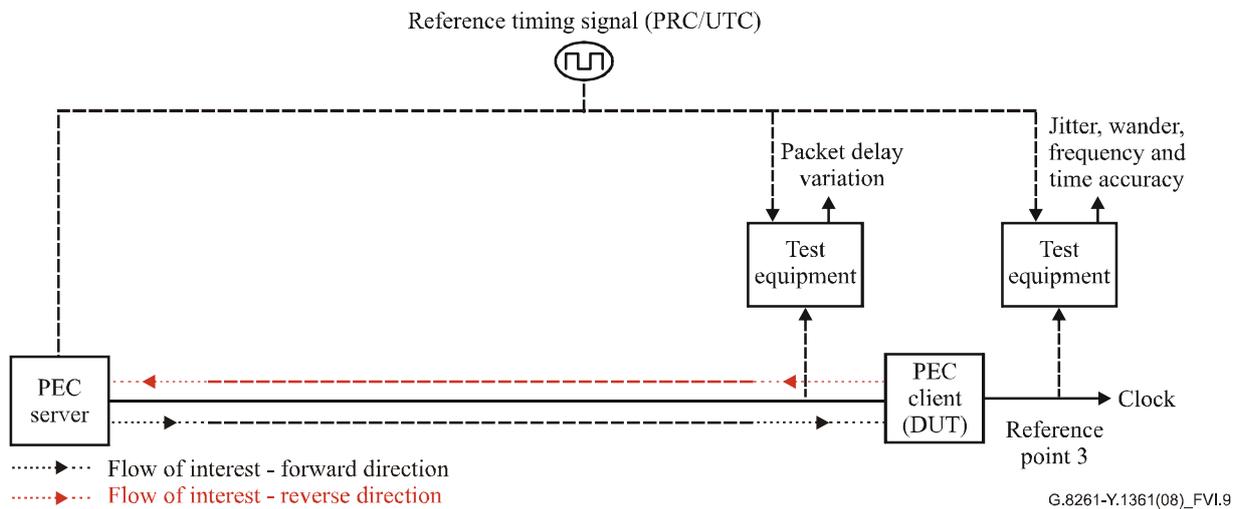


Figure VI.9 – Two-way baseline test topology

The baseline test should be done under the following conditions:

- No packet load
- Test measurements
 - Measure TIE, MTIE, and MRTIE (as described in [ITU-T G.823] and [ITU-T G.824])
 - Measure frequency accuracy (the value for the frequency accuracy measurement integration-time is dependent upon the relevant end equipment)
 - Measure peak-to-peak TOD accuracy
 - Performance should meet the network limits for the relevant cases as defined in clause 9.

VI.5.2 Performance test

The performance test is equivalent to Model A in Appendix V, composed with either 10 gigabit Ethernet switches or 9 gigabit Ethernet switches and 1 fast Ethernet switches. The test topology is shown in Figure VI.10.

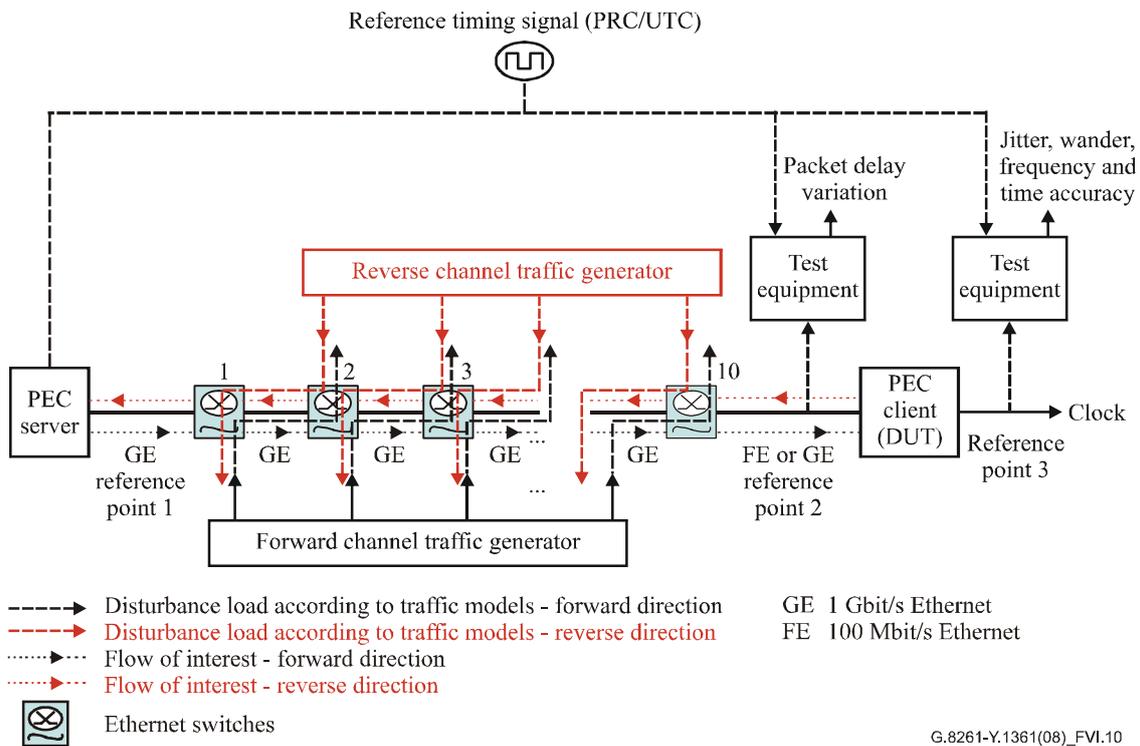


Figure VI.10 – Performance test topology for testing two-way protocols

The DUT must be tested for stability of operation under disruptive events that may cause the synchronization to fail or go out of specification. Test Cases in this clause are performed to test the DUT under load variation, network changes and packet loss.

For each of the test cases described in this clause, the following measurements should be performed:

- Measure TIE, MTIE, and MRTIE (as described in [ITU-T G.823] and [ITU-T G.824])
- Measure frequency accuracy (the value for the frequency accuracy measurement integration-time is dependent upon the relevant end equipment)
- Measure packet delay variation
- Measure peak-to-peak TOD accuracy
- Performance should meet the network limits for the relevant cases as defined in clause 9.

NOTE 1 – The test set-up, as described in Figure VI.10, provides the starting point towards a common test scenario.

However, in order to get a test environment that will be simpler to be implemented, and in order to remove any risk for getting different results when using Ethernet switches of different technologies, a proposal is under discussion to replace the specification as defined in Figure VI.10, with a new test set-up where in place of the Ethernet switches and the traffic generator, the delay variation could be created by a test equipment with a delay variation profile as input.

This delay variation profile could be expressed in terms of delay variation "test vectors" (test sequence) of duration 15 min, 60 min, and 24 hours. The delay variation shall be expressed with the proper timing resolution.

The test sequences would be based on the results from the tests performed using the tests topology as described in Figure VI.10.

NOTE 2 – Deterministic test cases may also be considered in addition to the test cases described in this clause. These are for further study.

VI.5.2.1 Input traffic characteristics

The same Traffic Models 1 and 2 defined in clauses VI.2.1 and VI.2.2 will be reused here for the two-way test cases.

NOTE – The definition of specific conditions to test for asymmetry is for further study. A simple test set-up could also consider constant delay in one direction.

VI.5.2.2 Test Case 12

Test Case 12 models the "Static" Packet load. Test Case 12 must use the following network conditions:

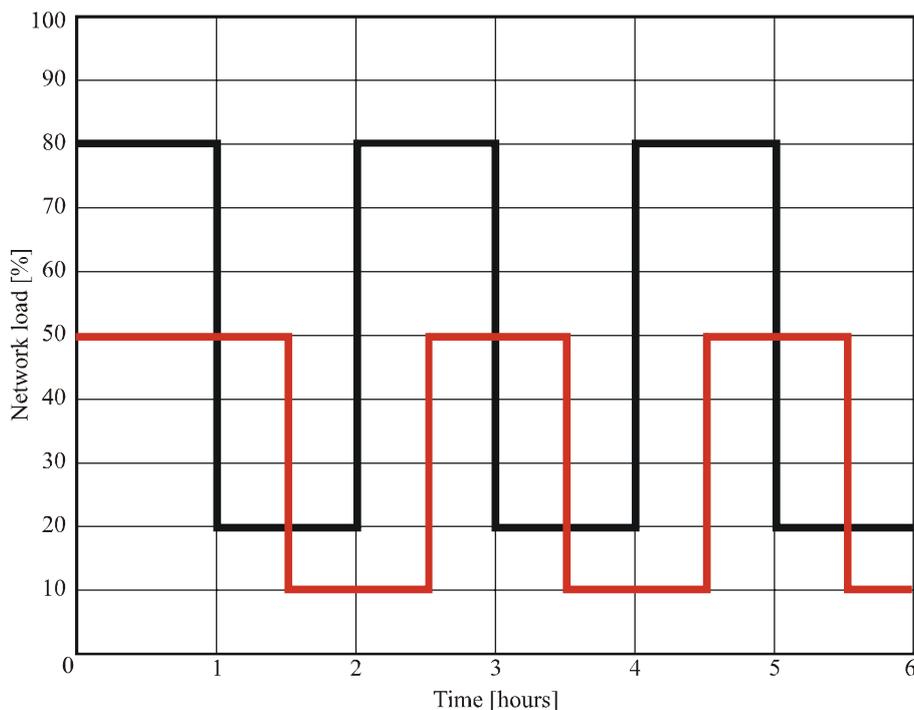
- Network disturbance load with 80% for the forward direction (Server to Client) and 20% in the reverse direction (Client to Server) for 1 hour. The test measurements should start after the clock recovery is in a stable condition. Guidance on stabilization period is provided in Appendix II. The disturbance background traffic to load the network must use Network Traffic Model 2 as defined in clause VI.2.2.

VI.5.2.3 Test Case 13

Test Case 13 models sudden large, and persistent changes in network load. It demonstrates stability on sudden large changes in network conditions, and wander performance in the presence of low frequency PDV.

Test Case 2 must use the following network conditions:

- The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
- Allow a stabilization period according to Appendix II for the clock recovery process to stabilize before doing the measurements.
- In the **forward** direction: Start with network disturbance load at 80% for 1 hour, drop to 20% for an hour, increase back to 80% for an hour, drop back to 20% for an hour, increase back to 80% for an hour, drop back to 20% for an hour. Simultaneously, in the **reverse** direction: Start with network disturbance load at 50% for 1.5 hours, drop to 10% for an hour, increase back to 50% for an hour, drop back to 10% for an hour, increase back to 50% for an hour, drop back to 10% for 0.5 hour (see Figure VI.11).



G.8261-Y.1361(08)_FVL11

Figure VI.11 – Sudden network disturbance load modulation for 2-way

- Repeat the test using the Network Traffic Model 2 as defined in clause VI.2.2 to load the network.

NOTE – The traffic generators in the test set-up are independent, because of this the shape of the traffic shown in Figure VI.11 may drift over time.

VI.5.2.4 Test Case 14

Test Case 14 models the slow change in network load over an extremely long timescale. It demonstrates stability with very slow changes in network conditions, and wander performance in the presence of extremely low frequency PDV.

Test Case 14 must use the following network conditions:

- The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
- Allow a stabilization period according to Appendix II for the clock recovery process to stabilize before doing the measurements.
- In the **forward** direction: Vary network disturbance load smoothly from 20% to 80% and back over a 24-hour period. Simultaneously, in the **reverse** direction: Vary network disturbance load smoothly from 10% to 55% and back over a 24-hour period (see Figure VI.12).

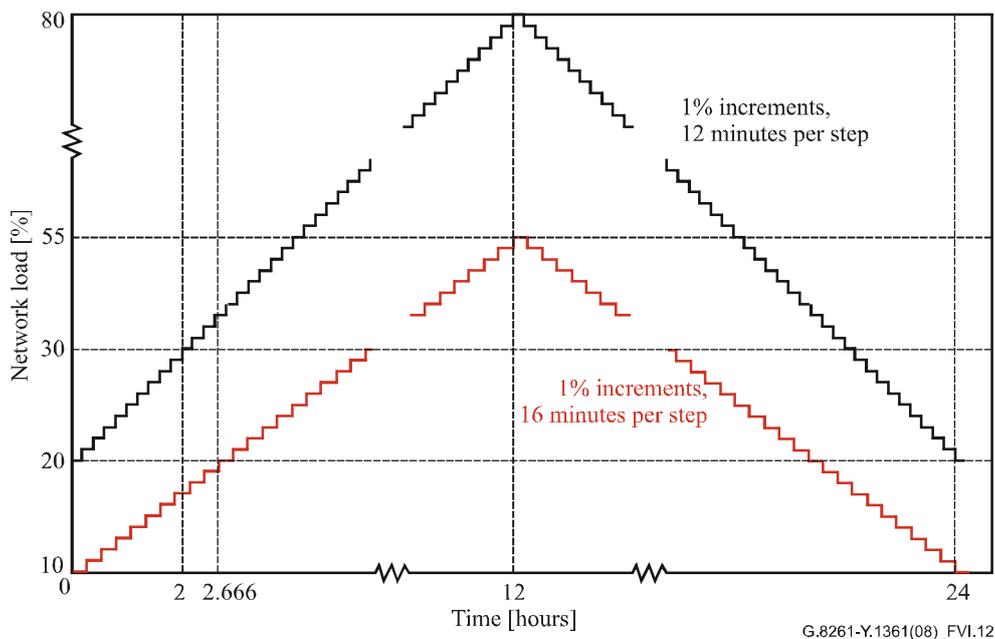


Figure VI.12 – Slow network load modulation for two-way

- Repeat the test using the Network Traffic Model 2 as defined in clause VI.2.2 to load the network.

VI.5.2.5 Test Case 15

Test Case 15 models temporary network outages and restoration for varying amounts of time. It demonstrates the ability to survive network outages and recover on restoration. It should be noted that MTIE over the 1000 s interruption will largely be governed by the quality of the local oscillator, and should not be taken as indicative of the quality of the clock recovery process.

Test Case 15 must use the following network conditions:

- The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
- Start with 40% of network disturbance load in the **forward** direction and 30% load in the **reverse** direction. After a stabilization period according to Appendix II, remove network connection for 10 s, then restore. Allow a stabilization period according to Appendix II for the clock recovery process to stabilize. Repeat with network interruptions of 100 s.
- Repeat the test using the Network Traffic Model 2 as defined in clause VI.2.2 to load the network.

VI.5.2.6 Test Case 16

Test Case 16 models temporary network congestion and restoration for varying amounts of time. It demonstrates the ability to survive temporary congestion in the packet network.

Test Case 16 must use the following network conditions:

- The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
- Start with 40% of network disturbance load in the **forward** direction and 30% load in the **reverse** direction. After a stabilization period according to Appendix II, increase network disturbance load to 100% in both directions (inducing severe delays and packet loss) for 10 s, then restore. Allow a stabilization period according to Appendix II for the clock recovery process to stabilize. Repeat with a congestion period of 100 s.

- Repeat the test using the Network Traffic Model 2 as defined in clause VI.2.2 to load the network.

VI.5.2.7 Test Case 17

Test Case 17 models routing changes caused by failures in the network.

Test Case 17 must use the following network conditions:

- Change the number of switches between the DUTs, causing a step change in packet network delay.
 - The packets to load the network must use Network Traffic Model 1 as defined in clause VI.2.1.
 - Start with 40% of network disturbance load in the **forward** direction and 30% load in the **reverse** direction. After a stabilization period according to Appendix II, re-route the traffic (in both directions) to bypass one switch in the traffic path. This shall be done by updating the test set-up in Figure VI.10 adding a cable from switch in position "n" to switch in position "n+2" and either using a fibre spool or adding an impairment box able to simulate different cable lengths (10 μ s and 200 μ s can be simulated as typical examples). The configuration shall be done so that the traffic flow under test is routed directly from switch in position "n" via the new link to switch in position "n+2".

After disconnecting the cable from switch "n" to switch "n+2" (so that traffic under test will then be routed from switch in position "n" to switch in position "n+1"), allow a stabilization period according to Appendix II for the clock recovery process to stabilize, and then reconnect the link that was disconnected in order to restore the traffic on the original path.
 - Start with 40% of network disturbance load in the **forward** direction and 30% load in the **reverse** direction. After a stabilization period according to Appendix II, re-route the traffic to bypass three switches in the traffic path. This shall be done by updating the test set-up in Figure VI.10 adding a cable from switch in position "n" to switch in position "n+4" and either using a fibre spool or adding an impairment box able to simulate different cable lengths (10 μ s and 200 μ s can be simulated as typical examples). The configuration shall be done so that the traffic flow under test is routed directly from switch in position "n" via the new link to switch in position "n+4".

After disconnecting the cable from switch "n" to switch "n+4" (so that traffic under test will then be routed from switch in position "n" to switch in position "n+1"), allow a stabilization period according to Appendix II for the clock recovery process to stabilize, and then reconnect the link that was disconnected in order to restore the traffic on the original path.
- Repeat the test using the Network Traffic Model 2 as defined in clause VI.2.2 to load the network.

Appendix VII

Wander limits in Deployment Case 1

(This appendix does not form an integral part of this Recommendation)

VII.1 Limits for the 2048 kbit/s interface

Table 1 has been calculated based on the following considerations with reference to Annex A in [ITU-T G.823].

The wander budget can be split into 3 main components:

- diurnal wander
- asynchronous mapping of 2048 kbit/s
- wander caused by clock noise and transients

Diurnal wander

There is no reason to change it, and its amplitude is small, 1 μ s.

Asynchronous mapping of 2048 kbit/s

An RMS law has been used to calculate the accumulation of the 2UI per island, 3 island will accumulate $\sqrt{3} * 2$ UI, i.e., 1.7 μ s, instead of 2 μ s in the original network model.

Wander caused by clock noise and transients

According to clause I.1.5 of [ITU-T G.823], the accumulation process may be different, depending on the magnitude of frequency offset, which may result in correlated or uncorrelated effects. It has been agreed an RMS noise accumulation. This means that each of the 4 islands is responsible for half of the wander budget, as currently indicated in this Recommendation. In the new network model, the 3 SDH islands are responsible for $\sqrt{3}$ of one SDH island budget according to the RMS accumulation law.

The total amount of wander allocated by [ITU-T G.823] is 15 μ s, and simulations reported 12.6 μ s.

The accumulation law between SDH and CES is different from that between SDH islands.

The noise generated in the SDH island is the result of VC-12 pointer events, which are infrequent, at least for a frequency offset in the range of 10^{-9} to 10^{-10} , as stated in clause I.1.5 of [ITU-T G.823]; this results in a very low probability that pointers occur at the same time in several islands.

As for the noise in a CES island, it looks very different from the one observed in SDH islands. This noise results from PDV.

As it has not been demonstrated that the RMS accumulation law applies between CES and SDH islands, it is proposed that the new model is assumed to have an RMS accumulation law for the 3 SDH island and a linear accumulation for the CES.

Thus, the wander budget that can be allocated to CES would be:

$$18 - (1(\text{diurnal wander}) + \sqrt{3} * 2\text{UI}(3 \text{ VC-12 mapping}) + 12.6/2 * \sqrt{3}(3 \text{ SDH islands})) = 4.3 \mu\text{s}$$

A wander of 4.3 μ s is then allocated to the CES for a period of 24 hours, and the wander template is reduced by a factor of 4.3/18 (0.24), for the other plateau derived from Table 2 of [ITU-T G.823].

VII.2 Limits for the 1544 kbit/s interface

The wander reference model and budget for 1544 kbit/s is specified in [ITU-T G.824] and consists of eight SDH islands. The wander budget components include switch synchronization, DS1 to DS3 mapping, DS1 to VC-11 mapping, Diurnal wander (Temperature effects on fibre), NE

synchronization noise and wander due to random pointers. The total budget of 18 microseconds (over 24 hours) allows 14.3 microseconds of wander between switches (refer to Figure A.1 of [ITU-T G.824]), and this has been subdivided to accommodate the replacement of one SDH island with a CES island. The procedure followed assumes that accumulation of mapping wander, synchronization noise and wander due to pointers is based on RMS addition. Based on RMS addition, the portion of the 18 microseconds available (i.e., 12.7, see Table VII.1) to each of eight islands is now 4.5 microseconds ($12.7/\sqrt{8}$).

Table VII.1 – 1544 kbit/s wander budget component allocation

Budget component	Allocation	Portion available for subdividing
Switch Synchronization	3.7	
E11-E31 mapping	0.3	
E11 to VC-11 mapping	2.6	2.6
Diurnal wander (Temp)	1.3	
NE sync noise/pointers	10.1	10.1
Total	18.0	12.7

The resulting wander for each island in terms of MTIE over all observation times up to 24 hours is given in Table 2. This table is based on a uniform reduction of the interface specification in Table 2 of [ITU-T G.824]. Note that this table also considers the mapping jitter requirements for a single VC-11 island, 0.7 UIpp as specified in [b-ITU-T G.783], (see Table 15-3 of [b-ITU-T G.783]).

The wander accumulation studies that were performed to derive the SDH wander components were based on extensive simulations to verify that the 18-microsecond requirement could be met over the SDH reference model. Future simulation work may be required when the CES network models and mappings are specified in greater detail. These numbers may be revised based on the results of that work.

Appendix VIII

Synchronization status messaging in synchronous Ethernet PHY

(This appendix does not form an integral part of this Recommendation)

Details on SSM for synchronous Ethernet are defined in [ITU-T G.8264].

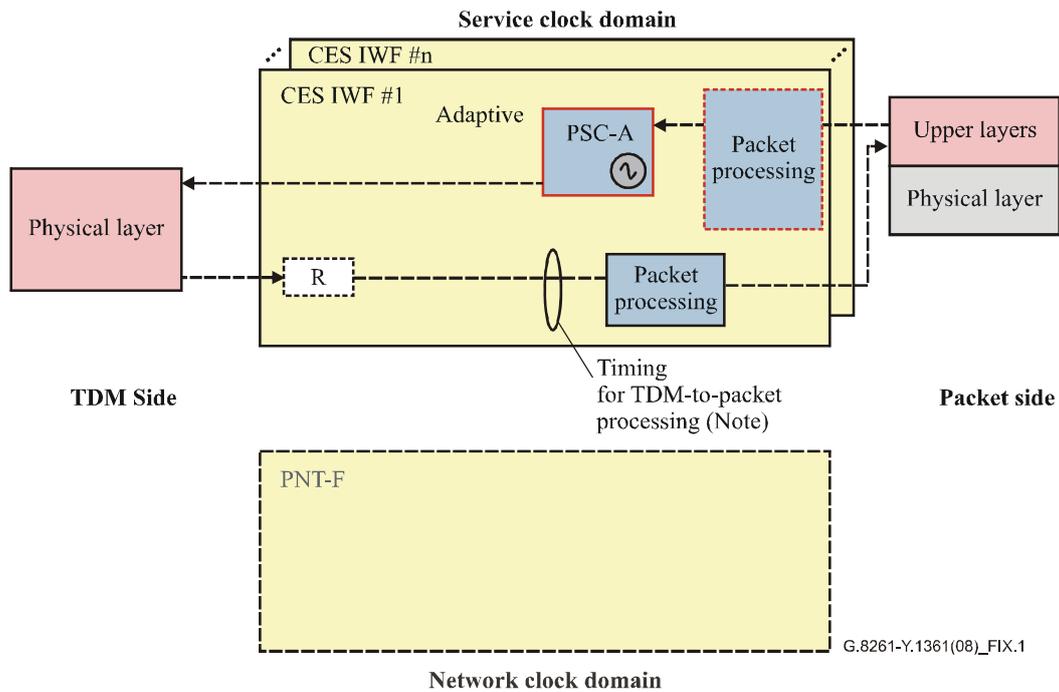
Appendix IX

IWF examples

(This appendix does not form an integral part of this Recommendation)

Examples of typical IWF applications are provided below.

Figure IX.1 shows the case when the service clock timing is handled via adaptive method and there is no PNT function implemented (no access to network clock).



-----> Timing flow

n Number of CES users terminated by the CES IWF

NOTE – This timing drives the generation of the outgoing packets and forms the basis for generating timing messages for adaptive methods.

Figure IX.1 – Adaptive method in the CES IWF; no PNT functions

The following figures present an example of the service and network domains in case of the TDM clock recovery according to the differential method where the common reference is distributed via synchronous Ethernet.

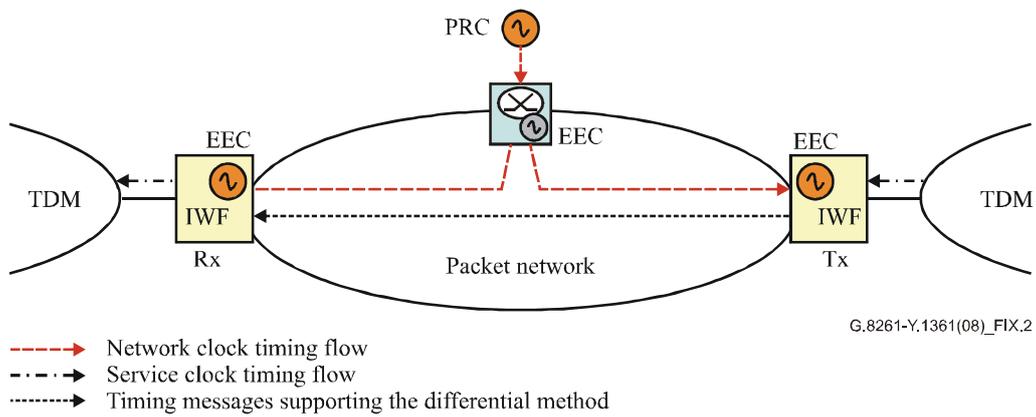


Figure IX.2 – Example of differential method using common reference distributed over synchronous Ethernet

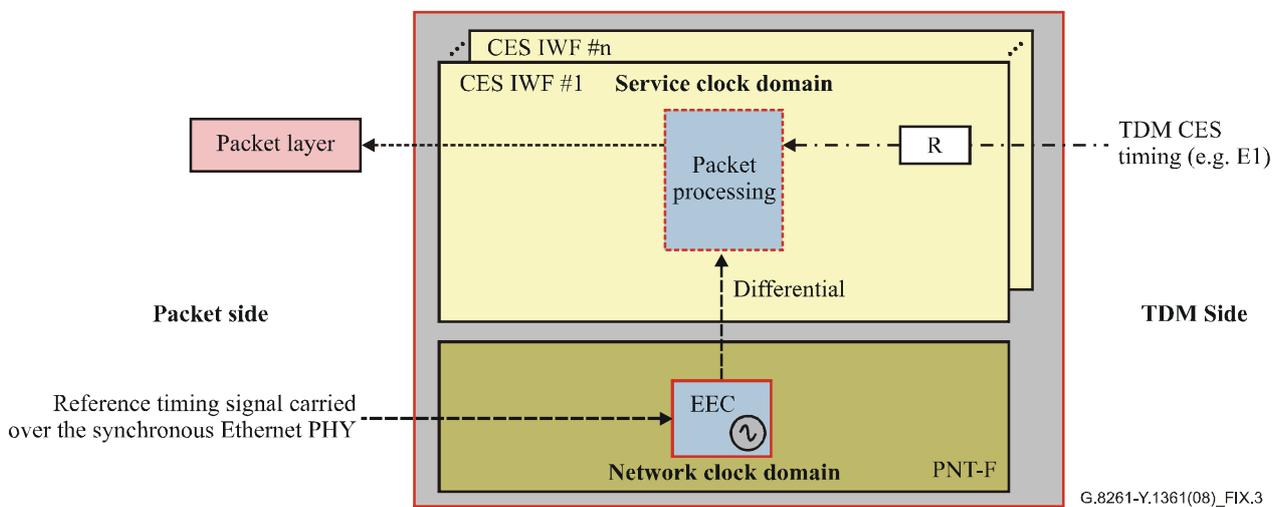


Figure IX.3 – Service and network clock domain in the IWF at the transmitting side (Tx)

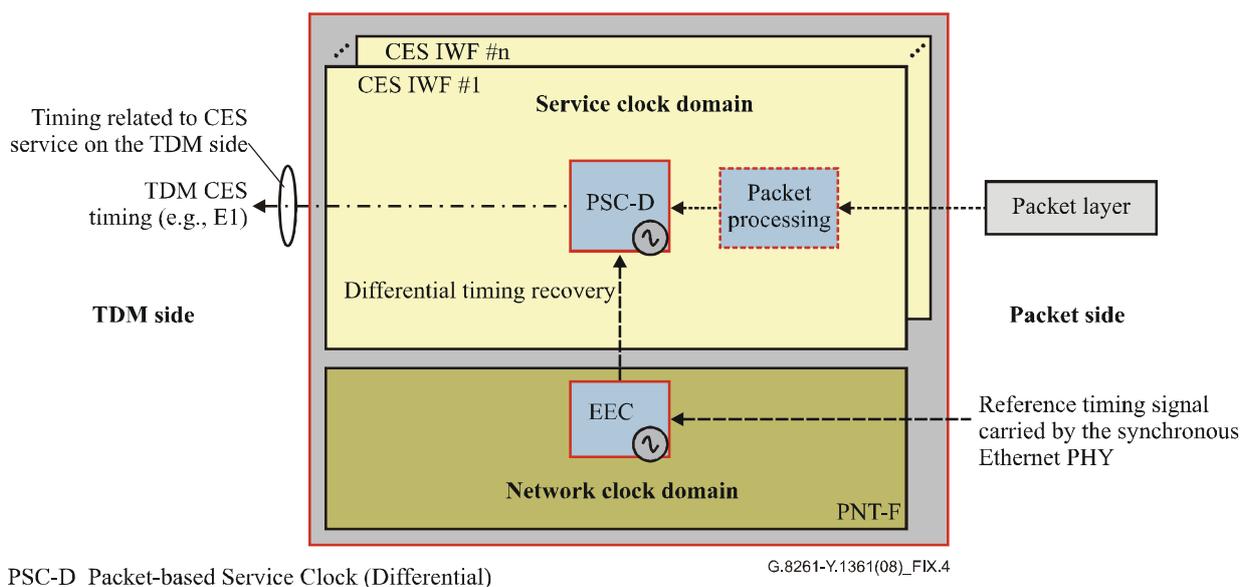
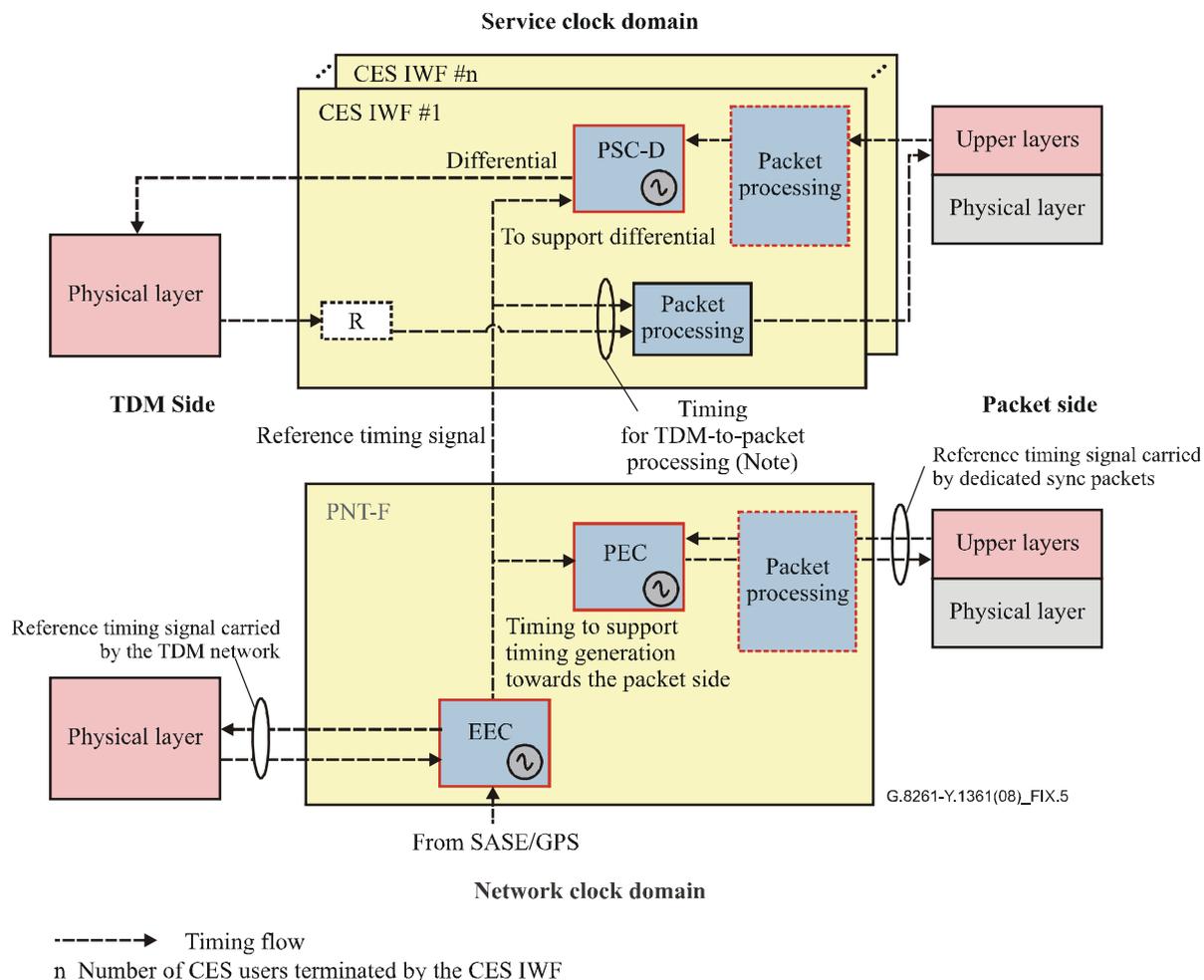


Figure IX.4 – Service and network clock domain in the IWF at the receiving side (Rx)

The next example shows the network timing carried by the TDM network. This timing is then used to support the differential operation in the CES IWF, and also to synchronize the PEC in order to generate the time stamps to be delivered over the packet network.



G.8261-Y.1361(08)_FIX.5

NOTE – This timing drives the generation of the outgoing packets and forms the basis for generating timing messages for the differential methods. The reference timing signal generated by the PNT-F and the one that shall support the differential method may be driven by a clock different from EEC (e.g., SEC).

Figure IX.5 – Differential method in the CES IWF; EEC and PEC in the PNT

Appendix X

Considerations on measurement of synchronous Ethernet according to ITU-T methodologies in comparison with IEEE jitter measurements

(This appendix does not form an integral part of this Recommendation)

The specifications and test methodologies for jitter on Ethernet differ from those for SDH because different timing methods are used. In synchronous systems such as SDH, the system components are synchronized to a common clock. In asynchronous systems such as Ethernet, component timing is provided either by distributed clocks or by clock signals recovered from the data. In this case, the jitter generated by components must be limited, but the jitter transferred from one component to another is less important than for synchronous systems where jitter can increase from component to component.

In SDH systems, three relevant measurements in different test configurations define jitter performance: band-limited jitter generation, sinusoidal jitter input tolerance, and jitter transfer.

Ethernet uses the approach that there are essentially two mechanisms that cause jitter, namely deterministic jitter and random jitter. Separate requirements are specified for transmitters and receivers.

Table X.1 – Comparison between ITU-T and IEEE jitter measurements

	SDH	Ethernet
Network standard	[b-ITU-T G.783], [ITU-T G.825]	[IEEE 802.3]
Test equipment standard	[ITU-T O.172]	
Jitter applications	Jitter generation Jitter input tolerance Jitter transfer	(Note 1) (Note 2) –
NOTE 1 – There are three viable methodologies for measuring jitter output: 1) Time domain measurement using an oscilloscope to characterize the data eye. 2) Time domain measurement using BERT scan by moving of the data sampling point within the data eye. 3) Time interval analysis based on accurate measurement of the time interval between threshold crossings of the transmitter waveform. NOTE 2 – A stressed receiver sensitivity (SRS) test is performed on receivers. The test is designed to verify that a receiver can operate at a BER of better than 10^{-12} when receiving the worst-case permitted signal. This test is analogous to jitter tolerance. The SRS test is also called a "stressed eye test" or "receiver tolerance test". A SRS test consists of two parts: an eye mask with combinations of impairments and a sinusoidal jitter template used for step-through measurements.		

Appendix XI

Relationship between requirements contained in this Recommendation and other key synchronization related Recommendations

(This appendix does not form an integral part of this Recommendation)

ITU-T has approved a family of Recommendations (G-series), which describe several aspects of synchronization functions for TDM. They are:

- G.803 – Architecture of transport networks based on the synchronous digital hierarchy (SDH) – This Recommendation describes the functional architecture of transport networks, including network synchronization principles for networks that are based on the SDH.
- G.810 – Definitions and terminology for synchronization networks – This Recommendation provides definitions and abbreviations used in timing and synchronization Recommendations.
- G.823 – The control of jitter and wander within digital networks which are based on the 2048 kbit/s hierarchy – This Recommendation specifies the maximum network limits of jitter and wander that shall not be exceeded, and the minimum equipment tolerance to jitter and wander that shall be provided at any relevant transport or synchronization interfaces which are based on the 2048 kbit/s hierarchy.
- G.824 – The control of jitter and wander within digital networks which are based on the 1544 kbit/s hierarchy – This Recommendation specifies the maximum network limits of jitter and wander that shall not be exceeded at relevant transport or synchronization network interfaces, and the minimum equipment tolerance to jitter and wander that shall be provided at any relevant synchronization or transport interface.
- G.825 – The control of jitter and wander within digital networks which are based on the synchronous digital hierarchy (SDH) – This Recommendation specifies the maximum network limits of jitter and wander that shall not be exceeded, and the minimum equipment tolerance to jitter and wander that shall be provided at any relevant transport or synchronization interfaces which are based on the synchronous digital hierarchy (SDH).
- G.812 – Timing requirements of slave clocks suitable for use as node clocks in synchronization networks – This Recommendation outlines minimum requirements for timing devices used as node clocks in synchronization networks. This Recommendation includes specifications for three types of clocks in the main body and three other clocks in Annex A.
- G.813 – Timing characteristics of SDH equipment slave clocks (SEC) – This Recommendation outlines minimum requirements for timing devices, used in synchronizing network equipment, that operate according to the principles governed by the synchronous digital hierarchy (SDH).
- G.781 – Synchronization layer functions – This Recommendation defines the atomic functions that are part of the 2 synchronization layers, the synchronization distribution (SD) layer and the network synchronization (NS) layer. It also defines some atomic functions, part of the transport layer, which are related with synchronization.
- G.783 – Characteristics of synchronous digital hierarchy (SDH) equipment functional blocks – This Recommendation specifies both the components and the methodology that should be used in order to specify SDH functionality of network elements.

ITU-T has been working on a family of Recommendations (G.826x- and Y.136x-series), which describe several aspects of synchronization functions for packet networks. They are:

- G.8261/Y.1361 – Timing and synchronization aspects in packet networks – This Recommendation defines synchronization aspects in packet networks. It specifies the maximum network limits of jitter and wander that shall not be exceeded. It specifies the minimum equipment tolerance to jitter and wander that shall be provided at the boundary of these packet networks at TDM and synchronization interfaces. It also outlines the minimum requirements for the synchronization function of network elements.
- G.8262/Y.1362 – Timing characteristics of synchronous Ethernet equipment slave clock (EEC) – This Recommendation outlines the requirements for timing devices used in synchronizing network equipment that uses synchronous Ethernet.
- G.8264/Y.1364 – Timing distribution through packet networks – This Recommendation outlines the requirements on Ethernet networks with respect to frequency transfer. It specifies the SSM transport channel namely the Ethernet synchronization messaging channel, protocol behaviour and message format. This Recommendation also details the required architecture in formal modelling language.

Additional work is ongoing at ITU-T to develop new Recommendations addressing other aspects of synchronization in the packet network (e.g., clock specification for packet-based methods is planned).

Table XI.1 shows the relationship between the TDM synchronization Recommendation family and the packet synchronization Recommendation family.

Table XI.1 – TDM synchronization Recommendation family versus the packet synchronization Recommendation family

Requirements	TDM network	Packet network
Functional architecture and network synchronization requirements	G.803, G.810 G.823, G.824, G.825	G.8261/Y.1361
Equipment clock specification	G.812 (Type IV), G.813	G.8262/Y.1362
Synchronization layer functions, functional blocks, timing flow, and SSM	G.783, G.781	G.8264/Y.1364, G.781

Appendix XII

Basic principles of timing over packet networks

(This appendix does not form an integral part of this Recommendation)

XII.1 General

Consider the situation where a slave clock (aka client) derives its timing from a master clock (aka server). Packet exchanges between master and slave provide measurements of the transit delay and clock offset between the two. This is explained with respect to Figure XII.1. The principles of timing over packet networks described here are quite general. These are examples that are applicable to both one-way and two-way methods. The particular protocol (e.g., [b-IEEE-1588] or NTP) employed determines the details (method, and underlying conventions), whereby the measurements ("time-stamps") are communicated between the two entities. It should be noted that the number of packets transmitted in the two directions need not be equal and, further, there may be additional packets transmitted that carry information but whose transit delay is not measured.

A fundamental assumption is that packet paths (routes) can be viewed as static over some interval of time, with fundamental changes occurring infrequently. If the time interval between significant changes of the transmission path is much larger than the packet exchange interval, the path can be treated as constant for a given set of measurements. That is, the path taken by the packets is the same over the measurement interval.

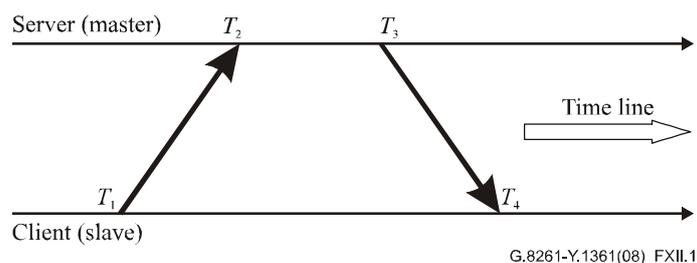


Figure XII.1 – Notion of time-stamps in packet exchange between a server and a client

Associated with the packets whose transit time is measured, there are four time-stamps which are defined as follows:

- T_1 : A time-stamp representing the best estimate of the transmit origination epoch of a packet or frame originating at the slave clock.
- T_2 : A time-stamp representing the best estimate of the receive termination epoch of packet or frame terminating at the master clock.
- T_3 : A time-stamp representing the best estimate of the transmit origination epoch of a packet or frame originating at the master clock.
- T_4 : A time-stamp representing the best estimate of the receive termination epoch of a packet or frame terminating at the slave clock.

A complete representation of a generic time-stamp value can be constructed as:

$$T_{TS}(n) = T(n) + e_{TS}(n) + e_{CLK}(n) \quad (\text{XII-1})$$

Equation XII-1 reflects the fact that the time-stamp (numerical value) associated with a packet, T_{TS} , is related to the true time epoch for that packet ($T(n)$) with two error terms. First, there is the direct contribution of the local clock error, e_{CLK} . Second, there is the inaccuracy in the time-stamp process,

e_{TS} , which can obscure the behaviour of the clock. The index " n " is included to identify the packet as being one member of a sequence of packets.

With reference to Figure XII.1, the following important timing flow metrics can be defined based on the time-stamps. These metrics are applicable in both one-way and two-way transfer operations.

First consider the case of one-way (frequency) transfer operation.

One-way transfer is an asymmetrical operation that only requires packet or PDU flow originating in one direction. For example, consider a timing flow originating at the master clock and terminating at the slave clock. Of the four time-stamps described in Figure XII.1, only two apply in this mode. When the master originates the time-stamp flow, convention dictates that the time-stamp pair (T_3 , T_4) describes the process. The originating time-stamp T_3 is with respect to the master view of time (Master time-scale), while the terminating time-stamp T_4 is with respect to the slave time-scale.

The measurement offset, δ_{MS} , can be calculated as (the subscript " MS " indicates the master-to-slave direction; " SM " is used for the slave-to-master direction):

$$\delta_{MS}(n) = T_4(n) - T_3(n) \quad (\text{XII-2a})$$

where:

$$T_4(n) = T(n) + \Delta_{MS}(n) + e_{S-TS}(n) + e_{S-CLK}(n) \quad (\text{XII-2b})$$

where $\Delta_{MS}(n)$ is the network delay experienced by the (n th) packet transmitted from the master to the slave, and:

$$T_3(n) = T(n) + e_{M-TS}(n) + e_{M-CLK}(n) \quad (\text{XII-2c})$$

Thus,

$$\delta_{MS}(n) = e_{S-CLK}(n) - e_{M-CLK}(n) + \Delta_{MS}(n) + e_{S-TS}(n) - e_{M-TS}(n) \quad (\text{XII-2d})$$

Note that one-way packet transfer between the client clock and server clock is also possible and an equivalent measurement offset defined for that case. Likewise, the measurement offset, δ_{SM} , for the slave-to-master direction, can be calculated as:

$$\delta_{SM}(n) = T_2(n) - T_1(n) \quad (\text{XII-2e})$$

and (Equation XII-2f) follows (Equation XII-2d) with the roles of master and slave reversed. $\Delta_{SM}(n)$ is the network delay experienced by the (n th) packet transmitted from the client to the server.

$$\delta_{SM}(n) = e_{M-CLK}(n) - e_{S-CLK}(n) + \Delta_{SM}(n) + e_{M-TS}(n) - e_{S-TS}(n) \quad (\text{XII-2f})$$

The most important properties of measurement offset are:

- 1) The measurement offset is biased by the one-way packet delay (Δ). The packet delay cannot be estimated with a one-way measurement, if the client clock offset is unknown. δ_{MS} and δ_{SM} are estimates of the one-way delay and are rendered imprecise by the clock and time-stamping errors.
- 2) By selecting one-way packet transactions with good (stable) delay properties, the deleterious impact of packet delay bias can be minimized.
- 3) The residual bias can either be reduced by estimating the one-way delay through some other means (such as using time-stamps associated with the reverse direction), or ignored in the case of frequency estimation as frequency offset is simply the rate of change of the phase offset which is zero for a constant phase bias error.
- 4) The time-stamping errors at the server and client cannot be eliminated and must be properly constrained for acceptable operation.

The measurement offset in a one-way packet transfer is analogous to the phase error measurements obtained with current physical layer one-way synchronization. As such, it is capable of supporting frequency transfer but not precise time transfer.

In contrast to one-way operation, two-way time-stamp operation implies timing packet flow in both directions. That is, all four time-stamps described in Figure XII.1 are employed. In a two-way packet time-stamp transaction, the time-stamp flow is initiated by one element (typically the client in NTP and the server in PTP).

The initiating direction is considered the forward direction, while the return transaction is considered the reverse direction. However, since each direction can be considered a one-way transaction, the two-way transaction can be described as follows:

$$\delta_{SM}(n) = e_{M-CLK}(n) - e_{S-CLK}(n) + \Delta_{SM}(n) + e_{M-TS}(n) - e_{S-TS}(n) \quad (\text{XII-3a})$$

$$\delta_{MS}(n) = e_{S-CLK}(n) - e_{M-CLK}(n) + \Delta_{MS}(n) + e_{S-TS}(n) - e_{M-TS}(n) \quad (\text{XII-3b})$$

Two key parameters can be estimated from the two-way exchange, namely from δ_{SM} and δ_{MS} . For simplicity, we shall assume for now that the time-stamping errors are negligible. The first key parameter is termed *offset*:

$$\text{offset}(n) = \frac{\delta_{MS}(n) - \delta_{SM}(n)}{2} = e_{S-CLK}(n) - e_{M-CLK}(n) + \frac{[\Delta_{MS}(n) - \Delta_{SM}(n)]}{2} \quad (\text{XII-4})$$

Where *offset* represents an estimate of the clock correction required to align the client time to the server time.

The second parameter is round-trip delay (*rtd*) which represents an estimate of the total round-trip path delay:

$$\text{rtd}(n) = \delta_{MS}(n) + \delta_{SM}(n) = \Delta_{MS}(n) + \Delta_{SM}(n) \quad (\text{XII-5})$$

Obviously, to obtain an unbiased offset estimate, the forward and reverse path delays must either be known or assumed symmetric. Note that an unbiased estimate of round-trip delay depends on the clock errors being the same for both directions. Of course, if the time between the two packet exchanges is low, then the clocks errors can be assumed common to both transactions.

Error in (the estimate of) the client clock, ϵ , can be attributed to the following causes:

- 1) The transit delay in the two directions is not equal. The difference directly affects the client clock error estimate. The error, $\Delta\epsilon$, is given by:

$$\Delta\epsilon = \left(\frac{1}{2}\right)(\Delta_{MS} - \Delta_{SM}) \quad (\text{XII-6})$$

- 2) The time-stamp measurements may not be measured precisely. That is, whereas T_1 is the actual time-of-departure of the packet from the server, the value used in the calculation may be an estimated time-of-departure. Likewise, T_2 is meant to be the actual time-of-arrival; the value used may be an estimate. For the time-stamp values to be accurate, they must be obtained by means that are as close to the PHY layer as possible and thus the time-of-departure (time-of-arrival) is not compromised by any (variable) delay attributable to such entities, as the operating system and interrupt handling. There will still be some residual errors associated with time-stamp resolution and delay variation in the PHY layer itself. Time-stamp resolution can be addressed by proper design. PHY noise needs to be either constrained or filtered depending on the transport.
- 3) The transit delays Δ_{MS} and Δ_{SM} are not fixed and change from packet to packet because of the packet delay variation (PDV) associated both with queuing related effects and physical transport effects in the network.

XII.2 Packet delay variation mitigation by packet selection

An important concept is that a clock filter or clock servo operating on the measurement parameters defined above may select or weight a transaction to optimize overall clock stability. That is, by suitable classification and selection of packets, the deleterious impact of packet delay variation can be mitigated.

The assumption that the path is constant over the interval of observation implies a situation where the packet delay variation will have a distribution function with a slowly changing floor. The floor is the minimum delay that a packet (or other protocol data unit such as a layer 2 frame) can experience in a given path. The floor can be viewed as the condition where both output and system queues (in all equipment that is involved in the flow, including the source, destination, and intervening elements) are "empty" when the particular packet needs the resource, and thereby do not delay transmission of the packet. The residual packet delay variation is then associated with physical layer jitter and wander mechanisms. Under normal non-congested loading conditions, in many cases, it has been observed that a reasonable fraction of the total number of packets will traverse the network at or near this floor, even though others may experience significantly longer delays. This type of behaviour has been addressed in this Recommendation (see Appendix I). In these cases, the PDV distribution exhibits a high degree of skewness when the network is lightly loaded. That is, the probability density can be more concentrated near this floor, with a relatively large fraction of total packets experiencing this "minimum" (or "near-minimum") delay. These phenomena are under study. A properly designed clock servo or filter can take advantage of the skewness to mitigate the effects of the instability on the long tail of the PDV distribution.

In principle, floor-based transfer noise is limited by a number of factors such as:

- 1) Physical layer propagation "speed of light" delay.
- 2) Time-stamp resolution
- 3) Mapping delays over non-Ethernet based physical transport (Ethernet over xDSL, PON, etc.)
- 4) Other small delay variation mechanisms, such a PHY clock jitter and backplane clock domain jitter.
- 5) Tilt in the offset of the local clock during the assessment of the floor.

XII.3 Comparison of packet-based and synchronous PHY methods

There are several differences between packet-based methods (e.g., [b-IEEE-1588], NTP) and synchronous PHY methods such as synchronous Ethernet. Some of these are discussed below.

- 1) Synchronous PHY methods are generally one-way methods and suitable for frequency alignment. Packet-based methods can be operated in a one-way mode to achieve frequency alignment and approximate time alignment. Packet-based operation in a two-way mode can achieve time alignment as well as frequency alignment.
- 2) Since the timing information in synchronous PHY methods is contained in the physical line-code signal, the information is not dependent on the traffic loading. On the contrary, packet-based methods are impacted by the traffic patterns, especially if quality-of-service prioritization schemes are not enforced.
- 3) Synchronous PHY methods are point-to-point. Every intermediate node between PRC and client clock under consideration must be part of the timing distribution system for timing chains to be unbroken. Packet-based methods can traverse nodes that are not involved in the timing distribution.
- 4) The input tolerance of a synchronous PHY clock is expressed in terms of the "clock noise" in the reference signal and quantified using TDEV and MTIE metrics. The network impairments that affect the performance of a packet-based clock are packet loss and packet delay variation (PDV) both from physical layer and queuing delay sources. Suitable metrics

for quantifying packet delay variation from a clock recovery point-of-view are in development. These include TDEV and minTDEV. MTIE is not a meaningful metric for quantifying packet delay variation from the viewpoint of clock recovery because not all packets are necessarily utilized in the recovery algorithms.

XII.4 Existing standards

The published standards for synchronization over packet networks are NTP (see [b-IETF RFC 1305]), [b-IEEE-1588] (PTP) and SNTP (see [b-IETF RFC 2030]). It should be noted that publication of Version 4 of NTP and Version 2 of PTP are planned for the same timeframe as the publication of this Recommendation.

NTP and PTP are general protocols for packet networks and do not directly address telecommunications requirements. Suitable profiles that provide guidelines for deployment in telecommunications applications are in development.

[ITU-T Y.1731] utilizes time-stamps for establishing some performance criteria in Ethernet networks. It is instructive to see [ITU-T Y.1731] utilize exactly the 4 time-stamps as described here, and transports them in OAM frames between the two ends of a link.

Bibliography

- [b-ITU-T G.701] Recommendation ITU-T G.701 (1993), *Vocabulary of digital transmission and multiplexing, and pulse code modulation (PCM) terms.*
- [b-ITU-T G.707] Recommendation ITU-T G.707/Y.1322 (2000), *Network node interface for the synchronous digital hierarchy (SDH).*
- [b-ITU-T G.781] Recommendation ITU-T G.781 (1999), *Synchronization layer functions.*
- [b-ITU-T G.783] Recommendation ITU-T G.783 (2004), *Characteristics of synchronous digital hierarchy (SDH) equipment functional blocks.*
- [b-ITU-T G.801] Recommendation ITU-T G.801 (1988), *Digital transmission models.*
- [b-ITU-T G.810] Recommendation ITU-T G.810 (1996), *Definitions and terminology for synchronization networks.*
- [b-ITU-T G.1020] Recommendation ITU-T G.1020 (2006), *Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks.*
- [b-ITU-T I.363.1] Recommendation ITU-T I.363.1 (1996), *B-ISDN ATM Adaptation Layer specification: Type 1 AAL.*
- [b-ITU-T T4] Recommendation ITU-T T.4 (2003), *Standardization of Group 3 facsimile terminals for document transmission.*
- [b-ITU-T V.90] Recommendation ITU-T V.90 (1998), *A digital modem and analogue modem pair for use on the Public Switched Telephone Network (PSTN) at data signalling rates of up to 56 000 bit/s downstream and up to 33600 bit/s upstream.*
- [b-ITU-T Y.1560] Recommendation ITU-T Y.1560 (2003), *Parameters for TCP connection performance in the presence of middleboxes.*
- [b-ETSI TR 101 685] ETSI TR 101 685 (in force), *Transmission and Multiplexing (TM); Timing and synchronization aspects of Asynchronous Transfer Mode (ATM) networks*
<http://webapp.etsi.org/workprogram/Report_WorkItem.asp?WKI_ID=7595>.
- [b-ETSI TS 100 594] ETSI TS 100 594 (1999), *Digital cellular telecommunications system (Phase 2+); Base Station Controller – Base Transceiver Station – (BSC – BTS) interface; Layer 1 Structure of Physical Circuits*
<http://webapp.etsi.org/workprogram/Report_WorkItem.asp?WKI_ID=8606>.
- [b-ETSI TS 125 104] ETSI TS 125 104 (in force), *Universal Mobile Telecommunications Systems (UMTS); Base Station (BS) radio transmission and reception (FDD)* <http://webapp.etsi.org/workprogram/Report_WorkItem.asp?WKI_ID=28301>.
- [b-ETSI TS 125 105] ETSI TS 125 105 (in force), *Universal Mobile Telecommunications Systems (UMTS); Base Station (BS) radio transmission and reception (TDD)* <http://webapp.etsi.org/workprogram/Report_WorkItem.asp?WKI_ID=28303>.
- [b-ETSI TS 125 402] ETSI TS 125 402 (in force), *Universal Mobile Telecommunications Systems (UMTS); Synchronization in UTRAN Stage 2*
<http://webapp.etsi.org/workprogram/Report_WorkItem.asp?WKI_ID=22972>.
- [b-ETSI TS 125 431] ETSI TS 125 431 (in force), *Universal Mobile Telecommunications System (UMTS); UTRAN Iub interface Layer 1*
<http://webapp.etsi.org/workprogram/Report_WorkItem.asp?WKI_ID=24642>.

- [b-ETSI TS 145 010] ETSI TS 145 010 (in force), *Digital cellular telecommunications systems (Phase 2+), Radio subsystem synchronization* <http://webapp.etsi.org/workprogram/Report_WorkItem.asp?WKI_ID=19334>.
- [b-IEEE 802.1ah] IEEE 802.1ah-2008, *IEEE Standard for local and metropolitan area network – Virtual Bridged Local Area Networks – Amendment 7: Provider Backbone bridges* <<http://www.ieee802.org/1/pages/802.1ah.html>>.
- [b-IEEE 802.1p] IEEE 802.1p-2005, *IEEE Standard for Local and Metropolitan Area Networks: Traffic Class Expediting and Dynamic Multicast Filtering*.
- [b-IEEE P802.1Qay] IEEE P802.1Qay-REV-2007, *Draft Standard for Local and Metropolitan Area Networks – Virtual Bridged Local Area Networks: Amendment Provider Backbone Bridge Traffic Engineering* <<http://www.ieee802.org/1/pages/802.1ay.html>>.
- [b-IEEE 802.3x] IEEE 802.3x-1997, *IEEE Standards for Local and Metropolitan Area Networks: Supplements to Carrier Sense Multiple Access with Collision Detection (CSMA/CD) Access Method and Physical Layer Specifications – Specification for 802.3 Full Duplex Operation and Physical Layer Specification for 100 Mb/s Operation on Two Pairs of Category 3 or Better Balanced Twisted Pair Cable* <<http://standards.ieee.org/getieee802/802.3.html>>.
- [b-IEEE 1588-PTP] IEEE 1588-PTP-2002, *IEEE Standard for a Precision Clock Synchronization Protocol for Networked Measurement and Control Systems* <<http://ieee1588.nist.gov/>>.
- [b-IETF RFC 1305] IETF RFC 1305 (1992), *Network Time Protocol (Version 3) – Specification, Implementation, and Analysis* <<http://www.ietf.org/rfc/rfc1305.txt?number=1305>>.
- [b-IETF RFC 2030] IETF RFC 2030 (1996), *Simple Network Time Protocol (SNTP) Version 4 for IPv4, IPv6 and OSI* <<http://www.ietf.org/rfc/rfc2030.txt?number=2030>>.
- [b-3GPP2 C.S0010-B] 3GPP2 C.S0010-B (in force), *Recommended Minimum Performance Standards for cdma2000 Spread Spectrum Base Stations* <http://www.3gpp2.org/Public_html/specs/C.S0010-B_v2.0_021704.pdf>.
- [b-3GPP2 C.S0002-C] 3GPP2 C.S0002-C (2002), *Physical layer standard for cdma2000 Spread Spectrum Systems* <http://www.3gpp2.org/Public_html/specs/C.S0002-C_v1.0.pdf>.
- [b-3GPP TR 25.836] 3GPP TR 25.836 (2001), *Node B synchronization for TDD* <<http://www.3gpp1.com/ftp/Specs/html-info/25836.htm>>.
- [b-MEF 3] MEF 3 (2004), *Circuit Emulation Service Definitions, Framework and Requirements in Metro Ethernet Networks* <<http://metroethernetforum.org/PDFs/Standards/MEF3.pdf>>.

ITU-T Y-SERIES RECOMMENDATIONS

GLOBAL INFORMATION INFRASTRUCTURE, INTERNET PROTOCOL ASPECTS AND NEXT-GENERATION NETWORKS

GLOBAL INFORMATION INFRASTRUCTURE	
General	Y.100–Y.199
Services, applications and middleware	Y.200–Y.299
Network aspects	Y.300–Y.399
Interfaces and protocols	Y.400–Y.499
Numbering, addressing and naming	Y.500–Y.599
Operation, administration and maintenance	Y.600–Y.699
Security	Y.700–Y.799
Performances	Y.800–Y.899
INTERNET PROTOCOL ASPECTS	
General	Y.1000–Y.1099
Services and applications	Y.1100–Y.1199
Architecture, access, network capabilities and resource management	Y.1200–Y.1299
Transport	Y.1300–Y.1399
Interworking	Y.1400–Y.1499
Quality of service and network performance	Y.1500–Y.1599
Signalling	Y.1600–Y.1699
Operation, administration and maintenance	Y.1700–Y.1799
Charging	Y.1800–Y.1899
NEXT GENERATION NETWORKS	
Frameworks and functional architecture models	Y.2000–Y.2099
Quality of Service and performance	Y.2100–Y.2199
Service aspects: Service capabilities and service architecture	Y.2200–Y.2249
Service aspects: Interoperability of services and networks in NGN	Y.2250–Y.2299
Numbering, naming and addressing	Y.2300–Y.2399
Network management	Y.2400–Y.2499
Network control architectures and protocols	Y.2500–Y.2599
Security	Y.2700–Y.2799
Generalized mobility	Y.2800–Y.2899

For further details, please refer to the list of ITU-T Recommendations.

SERIES OF ITU-T RECOMMENDATIONS

Series A	Organization of the work of ITU-T
Series D	General tariff principles
Series E	Overall network operation, telephone service, service operation and human factors
Series F	Non-telephone telecommunication services
Series G	Transmission systems and media, digital systems and networks
Series H	Audiovisual and multimedia systems
Series I	Integrated services digital network
Series J	Cable networks and transmission of television, sound programme and other multimedia signals
Series K	Protection against interference
Series L	Construction, installation and protection of cables and other elements of outside plant
Series M	Telecommunication management, including TMN and network maintenance
Series N	Maintenance: international sound programme and television transmission circuits
Series O	Specifications of measuring equipment
Series P	Telephone transmission quality, telephone installations, local line networks
Series Q	Switching and signalling
Series R	Telegraph transmission
Series S	Telegraph services terminal equipment
Series T	Terminals for telematic services
Series U	Telegraph switching
Series V	Data communication over the telephone network
Series X	Data networks, open system communications and security
Series Y	Global information infrastructure, Internet protocol aspects and next-generation networks
Series Z	Languages and general software aspects for telecommunication systems