Specifications of the network connection points according to the European directive 1999/5/EC

FIXED NETWORK ACCESS TERMINATIONS

Part II
Analogue network terminations (PSTN)

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Disclaimer
KPN has been very careful in describing the specifications of her fixed network access terminations. The reader is encouraged to contact KPN in case of any doubt concerning the correctness of the content in this document or possible misinterpretations.

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1 General

1.1 Introduction

The subject of part II of "KPN Telecom, Fixed Network Access Terminations" is the telephone network infrastructure, the Public Switched Telephone Network (PSTN) of KPN Telecom.

The intention of the present part is to provide detailed information about the properties at the fixed analogue network interfaces, which are intended for connection of the customer's Terminal Equipment (TE); these interfaces are further called "PSTN Network Termination Points" (NTPs). Initially no line terminating equipment is applicable for the 2-wire analogue PSTN network termination; the NTP coincides with the Network Connection Point (NCP).
With the introduction of ADSL (Asymmetric Digital Subscriber Line) in combination with PSTN, the PSTN (narrow band) access capability is extended with a broadband access capability; i.e. with the use of filters, a broadband channel is created on the analogue 2-wire subscriber line above the frequency band, which is needed for the basic PSTN service. The narrow band PSTN access capabilities as described in the present part of the publication, is not changed except for the installation of the filter unit (often indicated as the POTS splitter) between the NCP and the NTP. Information on ADSL in combination with PSTN is provided in Part IV-A of the publication.

The Network Connection Point (NCP) is indicated by KPN Telecom as the IS/RA point (IS = Infrastructure; RA = Randapparatuur, Dutch for Terminal Equipment); the NCP is the point of separation of the responsibility of the network operator and the customer.
In case of ADSL over PSTN, a further separation of responsibility between the PSTN service provider and the broadband service provider can be distinguished; this is outside the scope of this Part of the publication.

The goal of the publication may be clear; it is important for both the network operator and its customers, that terminal equipment is available on the market, which can interwork with the network in an excellent way.

According to the European Telecommunication Regulation as being in force, i.e. Directive 99/5/EC, it is required that Public Network Operators (PNOs) publish there interface specifications in sufficient detail to permit the design of telecommunications terminal equipment, capable of utilizing all services provided through the corresponding interface.

The former regulations (i.e. Directive 91/263/EEC, in 1998 succeeded by Directive 98/13/EC) were based on a regime of compulsory essential requirements and mandatory type approval procedures. As a consequence of that regime, ETSI has produced standards for European harmonized attachment requirements for the PSTN, i.e. TBR 21, EN 301 437 and TBR 38. Further, ETSI has worked out and published some PSTN guidance documents, i.e. EG 201 120 and EG 201 121. See chapter 5 for more information about these ETSI documents.
This means that during this former regulatory regime much progress is made in European harmonization and standardization related to the

analogue PSTN. KPN intends to be in complete compliance with these standards and guides.
These ETSI documents, do not cover all aspects of interworking with the PSTN for all services and supplementary services, nor is interworking guaranteed in all circumstances. Additional information on services and the dynamic signalling and call handling procedures, as perceived at the analogue network terminations, is needed in order to fill up the gap.

With the present publication KPN intends to describe the technical properties at the Network Termination Points (NTPs) of KPN's PSTN in detail and so, intends to satisfy the publication requirements of Directive 99/5/EC.
The information in this publication has been edited carefully, and the contents will be updated regularly. KPN Telecom does welcome comments and suggestions for improvements of the publication.

1.2 Scope

The scope of the present PSTN part of "KPN Telecom: Fixed Network Access Terminations" is:
Defining the analogue PSTN Network Termination Points (NTPs) and providing detailed technical information about the properties at these NTPs for the benefit of TE manufacturers and suppliers to place excellent and qualified TEs on the market.

In order to place the information in the perspective of the network, some very general information about the PSTN and the provided services and facilities is also included. It is outside the scope of the publication to provide detailed information about the services and facilities; such information can be found in commercial product information as provided by KPN Telecom. The technical aspects of some facilities, are described in detail, because these facilities need special TE or special functions in TE to support them.
The former type approval requirements, which are contained in the ETSI documents, are related to a single item of equipment in a more or less static environment. In practice, TE has to function in a dynamic environment and in a multi-terminal structure at the customer's NTP. For giving insight in these matters, the dynamic call handling and signalling processes are described in detail and installation matters at the customer's premises are dealt with.

1.3 References

This section contains an overview of the relevant standards and documents to which reference is made in this part of the publication.

73/23/EEC
Low Voltage Directive (LVD Directive).

89/336/EEC
Directive on Electro-Magnetic Compatibility (EMC Directive).

91/263/EEC (edition 1991); 98/13/EC (edition 1998)
Directive on Telecommunication Terminal Equipment (the former TTE Directive).

99/5/EC
Directive on Radio equipment and Telecommunications Terminal Equipment and the mutual recognition of their conformity (the operative R&TTE Directive).

ETSI TBR 21*)
Terminal Equipment; Attachment requirements for pan-European approval for connection to the analogue PSTNs of TE (excluding TE supporting the voice telephony service) in which network addressing, if provided, is by means of DTMF signalling.

ETSI EN 301 437 *)(sometimes referenced to as TBR 37)
Terminal Equipment; Attachment requirements for pan-European approval for connection to the analogue PSTNs of TE supporting the voice telephony service in which network addressing, if provided, is by means of DTMF signalling.

ETSI TBR 38*)
PSTN; Attachment requirements for a terminal equipment incorporating an analogue handset function capable of supporting the justified case service when connected to the analogue interface of the PSTN in Europe.

*) ETSI deliverable under the former TTE Directive; under the operative R&TTE Directive to be seen as ETSI standard rather than as regulatory document.

ETSI EG 201 120
PSTN; Method of rating terminal equipment so that it can be connected in series and/or in parallel to a Network Termination Point (NTP).

ETSI EG 201 121
A guide to the application of TBR 21; Annex A: ATAAB Advisory Notes.

ETSI ES 201 235
Specification of Dual Tone Multi-Frequency (DTMF) Transmitters and Receivers;
Part 1: General
Part 2: Transmitters
Part 3: Receivers
Part 4: Receivers for use in Terminal Equipment for end-to-end signalling

ETSI EN 300 001
Attachments to PSTN; General technical requirements for equipment connected to an analogue subscriber interface in the PSTN.

ETSI EN 300 659-1, Annex B
PSTN; Subscriber line protocol over the local loop for display (and related) services; Part 1: On hook data transmission; Annex B (normative): DTMF based subscriber line protocol.

ETSI EN 300 778-1, Annex A
PSTN; Protocol over the local loop for display and related services;

Terminal Equipment requirements; Part 1: On-hook data transmission; Annex A (normative): DTMF based subscriber line protocol.

ITU-T Rec. E.180 / ITU-T Rec. Q.35
Technical characteristics of tones for the telephone service.

ITU-T Rec. E-series: Supplement No. 2
Various tones used in national networks.

ITU-T Rec. G.100-series
General characteristics of international telephone connections and circuits.

ITU-T Rec. G.114
One-way transmission time.

ITU-T Rec. G.117
Transmission aspects of unbalance about earth.

ITU-T Rec. G.131/G.164/G.165
Some recommendations related to stability, echo and echo control.

ITU-T Rec. G.763
Digital Circuit Multiplication equipment using 32 kbit/s ADPCM and digital speech interpolation.

ITU-T Rec. Q.551
Transmission characteristics of digital exchanges.

ITU-T Rec. Q.552
Transmission characteristics at 2-wire analogue interfaces of digital exchanges.

ITU-T Rec. T-series
Terminal equipment and protocols for telematic services.

ITU-T Rec. V.25
Automatic answering equipment and general procedures for automatic calling equipment on the general switched telephone network including procedures for disabling of echo control devices for both manually and automatically established calls.

ITU-T Rec. V-series
Data communication over the telephone network.

T 11-series requirements (the former Dutch type approval requirements)
Conformity specification for terminal equipment intended for connection to the Public Switched Telephone Network in The Netherlands.

1.4 Abbreviations

This section contains the abbreviations used in this part of the publication, except those which are specific related to the detailed description of the analogue signalling; these are explained in the related chapters.

(AD)PCM	(Adaptive Differential) Pulse Code Modulation
AC	Alternating Current

ADSL	Asymmetric Digital Subscriber Line
ATAAB /AN	Analogue Type Approval Advisory Board / Advisory Note
CCBS	Completion of Calls to Busy Subscriber
CFU /B /NR	Call Forwarding Unconditional / Busy / No Reply
CgP /CdP	Calling Party / Called Party
CLIP	Calling Line Identification Presentation ("Nummerweergave")
CLIR	Calling Line Identification presentation Restriction ("Blokkering nummerweergave")
CPE	Customer's Premises Equipment
CWH	Call Waiting Hookflash ("Wisselgesprek®")
dBm	Logarithmic expression of Power, based on 1 milliwatt
dBV	Logarithmic expression of Voltage, based on 1 volt
DC	Direct Current
DCME	Digital Circuit Multiplication Equipment
DDI	Direct Dialling In feature
DSI /LRE	Digital Speech Interpolation / Low Rate Encoding
DTMF	Dual Tone Multi-Frequency
EG	ETSI Guide
EMC	Electro Magnetic Compatibility
EN /ES	European Norm / ETSI Standard
ETR	ETSI Technical Report
ETS	European Telecommunication Standard
ETSI	European Telecommunication Standards Institute
FTN	Forwarded To Number
GN	Group Number feature
IS	Infrastructure ("Infrastructuur")
ISDN	Integrated Services Digital Network
ITU-T	Telecommunication Standardization Sector of the International Telecommunication Union
LF /CF	ETSI defined Loading Factor / Dutch T 11 defined Connection Factor
LH	Line Hunting feature
LU_{ETSI} /LU_{NL}	Loading Unit: Arbitrary unit, ETSI defined / Dutch T 11 defined
LVD	Low Voltage Directive
MCID	Malicious Call Identification
NT	Network Termination
NCP/NTP/ TCP	Network Connection Point / Network Termination Point / Terminal Connection Point
PBX	Private Branch Exchange
PSTN/POTS	Public Switched Telephone Network / Plain Ordinary Telephone Service
P(T)NO	Public (Telecommunication) Network Operatorr
RR /HF	Register Recall signal / Hookflash signal
SC	Service Code
SLT /MLT	Single-Line Termination / Multi-Line Termination
T 11	The former National Technical Regulations for the PSTN in The Netherlands
TBR	ETSI Technical Base for Regulation
TCAM	Telecommunication Conformity Assessment and Market Surveillance Committee
TE /RA	Terminal Equipment / "Randapparatuur"
TTE/R&TTE	Telecommunications TE/Radio & Telecommunications TE

1.5 Legal regulations

The national telecommunication law and regulations have to be in conformity with the telecommunication directives as issued in the European Union.
Important for Public Telecommunication Network Operators (PTNOs) and Terminal Equipment manufacturers and suppliers is EC-Directive 99/5/EC, the Directive on Radio equipment and Telecommunication Terminal Equipment (R&TTE Directive); this R&TTE Directive is implemented in The Netherlands in the Telecommunication Law as well as in the Order in Council, known as the "Besluit Randapparaten en radioapparaten".
Also important are the Low Voltage Directive (LVD; 73/23/EEC), which is implemented by the Order in Council "Laagspanningsrichtlijn" and the EMC Directive 89/336/EEC (Electro Magnetic Compatibility), which is implemented by the Order in Council "EMC richtlijn".
More information with respect to the legal regulations related to telecommunication networks in The Netherlands can be obtained from the responsible government agency in The Netherlands.Also the web site of the OPTA (Dutch: Onafhankelijke Post en Telecommunicatie Autoriteit) can be consulted:

- http://www.opta.nl

The intention of the operative R&TTE Directive is, to create a situation in which new technological evolutions can be followed faster. A declaration by the manufacturer or his authorized representative, that the TE is intended for connection to a specified Network Termination Point (NTP) and complies with the related essential requirements and/or harmonized standards, will be sufficient to place the TE on the market.
In this situation, manufacturers or their authorized representatives shall have a greater liability for the functioning of their TEs.

An essential aspect in the R&TTE Directive, among other things, is article 4, "Notification and publication of interface specifications".
The Commission and its TCAM committee has provided the following guidance on these subject:

Guide 1: Guidance on interface notification by Member States
Guide 2: Guidance on interface publication by Public Telecommunications Network Operators
Guide 3: Guidance for Public Network Operators when publishing interfaces, and NRAs/Member States when supervising such publications
Guide 4: Commission guidance to terminal manufacturers and suppliers concerning interface publication

These guides can be downloaded from the web site:

- http://europa.eu.int/comm/enterprise/rtte/guides.htm

2 General description of the PSTN

2.1 Introduction

Originally the Public Switched Telephone Network or PSTN is designed for speech conversation between people over any distance, the switched telephony service. Switched means that a user, connected by means of an analogue access line to the PSTN, can use telephony Terminal Equipment to instruct the network to build up a connection to any other user irrespective whether that user is connected to the same network or to any other network with which interworking for the telephony or speech service is provided.

The PSTN is part of the fixed telecommunication network of KPN Telecom for narrowband switched services; a general description is provided in Part I, chapter 2.

In following sections, general information about the PSTN basic services and supplementary services is provided, and some transmission aspects of connections within the network of KPN Telecom are highlighted as introduction to the detailed information about the properties of the PSTN at the network termination points.
The present chapter does not contain the general description about signalling, dialling and call handling procedures; these are provided in subsequent chapters; e.g. chapter 3 contains the functional description of the PSTN call handling procedures.

2.2 Basic services and supplementary services

In this section general information related to the basic PSTN services and supplementary services (or facilities) is given and special aspects related to international connections are highlighted. It is not the intention to provide detailed information about all aspects of services, nor to present an exhaustive overview of all services and facilities, which are or can be delivered via the PSTN of KPN Telecom; such information can be found in commercial product information. Only facilities, for which support by Terminal Equipment (TE) is needed for its operation, are described in detail further on in this part of the publication; the related requirements are then provided.

2.2.1 Basic services

For many years in the long history of telephone networks, direct speech conversation between two customers, the voice telephony service, was the only relevant service of the PSTN. The PSTN, with its 300 - 3400 Hz band width connection capability, is designed for that task.

Today, voice telephony is still the elementary service in the PSTN for users to satisfy basic telecommunication needs; however, an increasing proportion of the traffic in the network is attributable to 'non-voice' services. For that reason, the PSTN can more general be indicated as the network for the 3,1 kHz voice band service (comparable with the 3,1 kHz audio bearer service in ISDN). Examples of 'non-voice' services are the information and data transport services by use of Terminal Equipment (TE) with the function of coding/decoding the information or data within the

3,1 kHz voice band, e.g. facsimile or fax apparatus according to ITU-T T-series Recommendations and modems according to ITU-T V-series Recommendations; this type of 'non-voice' services is further in this section indicated as 'voice band data services'.

Except for end-to-end communication between customers, the capabilities of the PSTN are more and more used for access to dedicated services like access to Internet, Electronic mail and voice mail services, access to information providers (e.g. calls to 0800 and 090x (x=0,6,9) numbers), and so on.

With respect to the performance of the PSTN for 'voice band data services', this is very dependent of the network connection, e.g. the length and quality of analogue sections, the number of analogue to digital (A/D) conversions, the modem type, the traffic handling capability in other networks, e.g. the network of a special service provider.
In the PSTN of KPN Telecom, only the access lines and eventual a part of the local exchanges are analogue; the trunk network is completely digital. It can be expected that high performance voice band data connections are possible within the network of KPN Telecom, e.g. the high bit rates as defined for V.32 or V.34 type of modems, are very well possible. And if the other side is digital connected, which is normal practice at present for access to Internet, very high bit rates with V.90/V.92 modems are common use.
For international connections, a concatenation of analogue and digital sections may degrade the 'voice band data service'. A decrease in the performance of modems may also be caused by restrictions in the traffic handling capability of systems, e.g. the availability 40 kbit/s capabilities in DCME systems (see hereafter) for dedicated 'voice band data services' during busy hours.

In case of international connections over long distances, e.g. transoceanic cable systems and satellite systems, the communication performance may be influenced by special transmission facilities in the connection.
The following aspects are highlighted hereafter:

- One way propagation time;
- Echo control devices;
- Digital Circuit Multiplication Equipment (DCME).

- ***One way propagation time***

On long international connections, the propagation time causes signal delays which are not negligible and may decrease the performance of the communication.
As far as the voice telephony service is concerned, the delay over the longest terrestrial connections via transoceanic cable are, in general, not experienced as inconvenient for the communication. If a satellite system, with its one way propagation time of about 270 ms, is contained in the connection, the influence of the signal delay is annoying.
ITU-T recommends that, for speech connections in normal cases, one way propagation times of more than 400 ms should be avoided. This means that more than one satellite hop in a connection should be avoided. This is not always possible; connections via communication satellites are not only applied on international routes, but also on domestic routes for reaching difficult accessible regions in countries. If it is possible, on the basis of information provided by such countries, to determine in the international

gateway exchanges of KPN Telecom, that at the destination side a domestic satellite hop shall be contained in the connection, the call will be routed via a terrestrial route to the destination country. E.g. for some regions in Indonesia this is the case; and also traffic to the Inmarsat satellite for the Pacific Ocean region is routed over terrestrial links to an earth station which has full sight with the related satellite.
Further information on this subject can be found in e.g. ITU-T Rec. G.114.

- ***Echo control devices***

For voice telephony connections with a significant one way propagation time, e.g. long distances via cable routes or satellite routes, echo control devices are necessary for an acceptable quality level of the speech conversation.
These echo control devices are not desirable in modem connections; they may degrade the performance of 'voice band data services'. Most of the V-series modems and T-series fax apparatus use the 2100 Hz answering tone as defined in ITU-T Recommendation V.25. If, at call establishment, this tone is detected, the echo control devices in the connection are disabled for the duration of the call.
Further information on this subject can be found in e.g. ITU-T Recs. G.131, G.164 and G.165.

- ***Digital Circuit Multiplication Equipment (DCME)***

On long cable connections, e.g. transatlantic cable systems, and via satellite systems, special equipment is in use for increasing the efficiency of the transmission system. Such equipment is designed and optimized for the voice telephony service. In digital transmission techniques Digital Circuit Multiplication Equipment (DCME) (see e.g. ITU-T Rec. G.763) is applied.
The techniques, utilised in DCME for a more efficient use of international transmission means, are Digital Speech Interpolation (DSI) and Low Rate Encoding (LRE).
DSI makes use of the fact that, in an average speech conversation, there is only about half of the time speech activity in one or in the other direction; only during speech activity a speech path is made available in the related direction.
LRE is a voice band signal encoding method, e.g. Adaptive Differential Pulse Code Modulation (ADPCM), which results in a bit rate less than the 64 kbit/s as for standard Pulse Code Modulation (PCM), e.g. 40 kbit/s, 32 kbit/s, 24 kbit/s. LRE techniques give an acceptable performance for the speech telephony service, but LRE with bit rates below 40 kbit/s may be problematic for 'voice band data services'.
As indicated before for the echo control device case, most of the V-series modems and T-series fax machines use the 2100 Hz answering tone as defined in ITU-T Rec. V.25. If, at call establishment, this answering tone is detected in the DCME, the DCME will designate that call as a 'voice band data call' for which the bit rate shall be not less than 40 kbit/s.

2.2.2 Supplementary services

Supplementary services or facilities are provided by the PSTN in addition to the basic 3,1 kHz voice band service. These facilities can not be offered to a customer as a stand alone service; they are intended to supplement the basic service.

Some of the facilities can be provided in agreement with the customer on a subscription basis; other facilities are freely available. Facilities may be available for both Single-Line Terminations (SLT) and Multi-Line Terminations (MLT) or for network terminations of a dedicated kind only, e.g. only at a SLT. Details on the applicability and provisioning can be found in the product information.
In the following, short descriptions of a number of the facilities are given.

1) Call Forwarding (CF)

The CF supplementary service permits a customer to have the network sending all terminating calls, or just those terminating calls which meet certain conditions, to another destination. The customer's originating service is not affected.
On request from the customer, KPN can for the customer's access termination inhibit the incoming calls, which are forwarded due to this CF service.

The CF supplementary service can be activated by initiating a call and dialling the digits string:

∗<SC>∗<FTN>#

The activated CF supplementary service can be deactivated by again initiating a call and dialling the digit string:

#<SC>#

If the activation or the deactivation is successful, acceptance tone (same characteristic as dial tone) will be returned; else, rejection tone (same characteristic as congestion tone) is returned.

The meaning of <SC> and <FTN> are:

<SC>: Service Code; a 2-digit code which identifies the CF version;
<FTN>: Forwarded To Number; the number to which the call has to be forwarded.

Different versions of CF are defined; each version is identified with its own Service Code (SC):

Call Forwarding Unconditional (CFU):	<SC> = 21
Call Forwarding Busy (CFB):	<SC> = 67
Call Forwarding No Reply (CFNR):	<SC> = 61

With CFU activated, all calls will be forwarded irrespective of the condition of the termination line.
With CFB activated, the calls, meeting the termination line 'busy', will be forwarded.
With CFNR activated, the terminated calls, which are not answered within about 20 seconds (4 to 5 ringing cycles), will be forwarded.

When a call is initiated at a network termination, for which CFU is activated, a special dial tone is provided in stead of the normal continuous dial tone; the special dial tone is the same as the normal dial tone with short interruptions every 500 ms. The intention of the special dial tone is to remind the user that, because of the activated CFU, the network termination can not be reached. So TE, intended for connection to network terminations on which CFU is applicable and for which dial tone detection is required, should support the detection of the special dial tone.

On an SLT, all CF services are freely available.
On an MLT with maximum 6 lines and with the group number feature (see the related section in chapter 4), CFU can be provided operator controlled, either on subscription base or as a temporary solution in case of infrastructure faults.

KPN Telecom is providing a "Voice mail" service with the use of the CFNR supplementary service to the Voice mail number <0842 333>, i.e. "voice mail" is activated with **<∗61∗0842333# >** and deactivated with **<#61#>**.

2) Call Waiting Hookflash (CWH)

The CWH supplementary service (Dutch: 'Wisselgesprek®') can be provided to customers with an SLT.
CWH permits the customer to have the network, during an active call, indicating a new incoming call to the user; and further, with the use of a hookflash signal, to answer the new call and subsequently to switch between one and the other call. The new call is indicated by a special tone, the 'call waiting tone'.
CWH can be activated permanently by KPN Telecom in agreement with the customer. The customer can deactivate CWH on a per call basis. If a customer, with an activated CWH supplementary service, originates a call during which no indication of a new call is wanted, CWH can be deactivated for the duration of that call by dialling the digit string:

#43∗<CdPN>

In this string, <CdPN> is the called party's telephone/ISDN number for establishing the call.
Further information on CWH is provided in a related section in chapter 6.

3) Calling Line Identification Presentation (CLIP) and CLI Restriction (CLIR)

The CLIP supplementary service (Dutch: 'Nummerweergave') can be provided to customers with an SLT.
CLIP provides the called party with the possibility to receive the calling party's telephone/ISDN number (CgPN). CLIP can be activated permanently by KPN Telecom in agreement with the customer.
Because special TE (or special functionality in TE) is required for the support of CLIP, detailed technical information is provided in chapter 6.

The CLIR supplementary service (Dutch: 'Blokkering Nummerweergave') enables the calling party to prevent presentation of the customers telephone/ISDN number to the called party. The default setting of CLIR is 'not activated', which means that presentation of the number is allowed.
In agreement with the customer, CLIR can be activated permanently by KPN Telecom, which means that then presentation of the number is not allowed.
For the case of CLIR is not activated, the user can on a per call basis restrict the presentation by dialling the digit string:

∗31∗<CdPN> or **131<CdPN>**

In this string, <CdPN> is the called party's telephone/ISDN number of the call.

4) Completion of Calls to Busy Subscriber (CCBS)

The CCBS supplementary service enables a calling user A, encountering a busy destination B, to have the call completed without having to make a new call attempt, when destination B becomes not busy.
KPN Telecom may restrict the destinations towards which the CCBS supplementary service can be used (e.g. international destinations, mobile destinations, other operators, 0800/0900-numbers). This restriction is part of the commercial offering of the service.
CCBS will only be applied and is generally available on SLTs.

i) CCBS activation
When receiving busy tone as the indication that destination B is busy, user A can activate CCBS by dialling the digit **<5>** during the time that the exchange is supplying busy tone (about 15 s), either with DTMF dialling or with decadic dialling.

- When the activation is successful, the CCBS request is registered in the network and acceptance tone (same characteristic as dial tone) is supplied.
- When the activation is unsuccessful, rejection tone (same characteristic as congestion tone) is supplied (dialling of an other digit than <5> while receiving busy tone, is interpreted as an unsuccessful CCBS activation).

In both cases, the tone is provided for maximum 5 s; when user A does not release within that time, the supplied tone indication will be changed to release tone (same characteristic as congestion tone) according to the normal release procedure.

A CCBS request will be registered in the network for a maximum of 45 minutes (the CCBS service duration timer is 45 minutes).

For a user A, a maximum of 5 CCBS requests can be registered simultaneously in the network.
For a destination B, also a maximum of 5 CCBS requests to that B-number can be registered simultaneously in the network.

In case user A requests activation of the CCBS supplementary service whilst an identical activation already exists, the new activation request will not be accepted.

ii) CCBS deactivation
For deactivation of registered CCBS request(s), the following control procedures apply (only possible with DTMF dialling):
a) **#37#** :deactivation for all outstanding CCBS requests at once;
b) **#37∗<TN>#** :deactivation for a specific outstanding CCBS request to destination B with telephone number TN.
In the latter case, <TN> shall be equivalent to the originally dialled B-number, i.e. including prefixes, but excluding possibly dialled supplementary service control digits (e.g. CLIR per call).

- When the deactivation is successful, the related CCBS request is erased in the network and acceptance tone (same characteristic as dial tone) is supplied.

– When the deactivation is unsuccessful, rejection tone (same characteristic as congestion tone) is supplied.

In both cases, the tone is provided for maximum 5 s; when user A does not release within that time, the supplied tone indication will be changed to release tone (same characteristic as congestion tone) according to the normal release procedure.

iii) CCBS interrogation

For interrogation of registered CCBS request(s), the following control procedures apply (only possible with DTMF dialling):

a) **∗#37#** :interrogation for all outstanding CCBS requests at once;

b) **∗#37∗<TN>#** :interrogation for a specific outstanding CCBS request to destination B with telephone number TN.

In the latter case, <TN> shall be equivalent to the originally dialled B-number, i.e. including prefixes, but excluding possibly dialled supplementary service control digits (e.g. CLIR per call).

– When the interrogation is successful, i.e. a) at least one CCBS request is registered or b) a CCBS request to <TN> is registered, acceptance tone (same characteristic as dial tone) is supplied.
– When the interrogation is unsuccessful, i.e. a) no CCBS request is registered or b) no CCBS request to <TN> is registered, rejection tone (same characteristic as congestion tone) is supplied.

In both cases, the tone is provided for maximum 5 s; when user A does not release within that time, the supplied tone indication will be changed to release tone (same characteristic as congestion tone) according to the normal release procedure.

iv) CCBS invocation and operation

Some 5 s after destination B has become not busy, a CCBS recall to user A will be made.

If user A is found busy, the CCBS recall is suspended during the time that the busy condition of A exists; user A will not be notified.

For the CCBS recall, normal ringing will be supplied; the <u>CCBS recall timer</u> is 20 s.

For a CCBS recall user A remains to be the originating side (calling line side) of the call. This means that services/facilities related to the originating side (e.g. metering, Outgoing Call Barring (OCB), Calling Line Identification Restriction (CLIR)) do apply; services/facilities related to the terminating side (e.g. Calling Line Identification Presentation (CLIP), Call Waiting Hookflash (CWH), call diversion services) do not apply.

For destination B, the call due to a CCBS recall from the A-side will be treated as a normal terminating call with all destination related services/facilities.

In case such a CCBS call meets destination B busy again, the CCBS request will not be retained by the network. In this case user A has to activate the service again.

User A and destination B can reside in different networks (e.g. operated by different operators) in case both networks and all intervening networks support the CCBS supplementary service.

5) *Metering*

Note:
The 50 Hz 'common mode' metering facility is no longer commercially available. The present version of the publication does contain the technical information about the facility, because the facility is still present in the network.

SLTs and MLTs can be provided with the metering facility.
The indications are given in the form of metering pulses per unit speech time (charging unit), in which the duration of a charging unit is dependent on the tariff for that call (e.g. some minutes for local calls, some seconds for international calls, a series of pulses at a rate of about 2 per second for special services). The cost for the call can be calculated by multiplying the number of pulses (charging units) with a fixed unit prize.
The first metering pulse is supplied at answer instance; this pulse can, except as a cost indication, also be used as an answer indication. It may be possible that for some services, for which the calling party is not to be charged, the first metering pulse is supplied with only the meaning 'answer indication'.
The metering pulses are supplied as 50 Hz pulses, sent in 'common mode' to the network termination provided with the metering facility; detailed technical information is contained in a related section in chapter 4.

6) *Outgoing call barring*

This facility permits a customer to have outgoing calls to certain destinations, originated at the customer's termination or invoked on his account by an activated supplementary service, to be rejected by the network. Examples are calls to certain 090x numbers, to international destinations, and such. The customer's terminating service is not affected. The facility can be provided in an agreement with KPN Telecom.

7) *Malicious Call IDentification (MCID)*

This facility can be provided by KPN Telecom on special request of a customer, e.g. in case of repeated unwanted frivolous calls. MCID enables a user to instruct the network that the source of an incoming call shall be identified and registered in the network.

2.3 Transmission aspects connections within network of KPN Telecom

2.3.1 Introduction

In the handling of telephone calls different conditions or phases can be identified, e.g. the idle condition, the dialling, ringing, communication and release phases; the call handling procedures of the PSTN are dealt with in chapter 3.
This section is related to the communication phase; the transmission properties of PSTN connections in the communication phase are described.

The digital exchanges in the telecommunication network of KPN Telecom are designed according to the ITU-T Rec. Q.500 series; the transmission characteristics are dealt with in the Rec. Q.550 series. For the transmission characteristics of the 2-wire analogue subscriber line interface of digital exchanges, the requirements for interface type Z, as specified in Q.551 and Q.552, are applicable.

But still a lot of subscriber lines terminate on an analogue local exchange or on an analogue switching stage of the local exchange. Further, some of the transmission properties of a connection are not only determined by the local exchange, but are also dependent on the other network elements and on the connected Terminal Equipment (TE). For that reason, some end-to-end transmission properties are described hereafter. In this description it is assumed that TE, complying with the former technical regulations, is connected to both the originating and terminating Network Termination Points (NTPs).
In this description worst case values are presented which gives an impression of the minimum quality, which can be expected for a connection under normal circumstances. In most cases the properties of a connection will be much better.

The figures in this section apply for national connections within the switched telecommunication network of KPN Telecom. General characteristics of international telephone connections and circuits can be found in the ITU-T G.100 series recommendations.

2.3.2 Transmission aspects of analogue terminated PSTN connections

In this subsection the transmission properties of a PSTN connection in communication phase between two analogue NTPs of the PSTN of KPN Telecom are provided.

The following parameters are dealt with:
1) Transmission loss;
2) Loss distortion with frequency;
3) Group delay distortion with frequency;
4) Noise;
5) Variation of loss with input level;
6) Discrimination against out-of-band signals;
7) Total distortion, including quantizing distortion
8) Impedance;
9) Crosstalk;
10) Spurious signals.

1) Transmission loss

The transmission loss of a telephone connection between two analogue NTPs in the PSTN of KPN Telecom is maximum 22 dB for an input signal with a frequency of 1020 Hz and a level of -10 dBm.
The minimum value of the transmission loss can be 0 dB.

2) Loss distortion with frequency

The loss distortion with frequency of a telephone connection between two analogue NTPs is within the mask of figure II-2/1. This is valid for an input level of -10 dBm and with the transmission loss at 1020 Hz as reference.

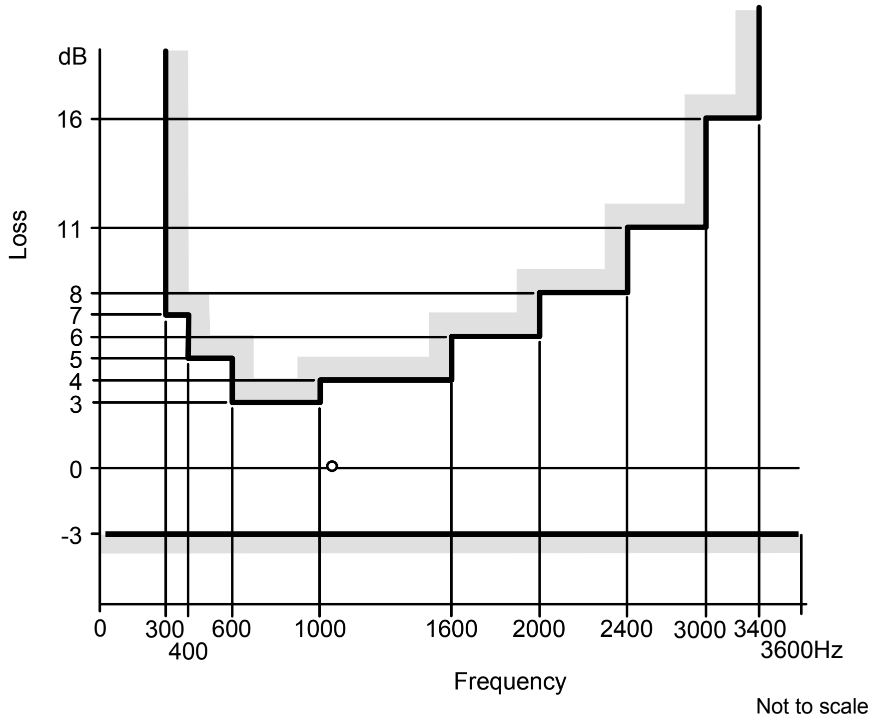


Figure II-2/1: Loss distortion with frequency

3) Group delay distortion with frequency

The maximum group delay distortion of a telephone connection between two analogue NTPs is provided in table II-2/1. The reference is the minimum group delay in the frequency range 300 - 3400 Hz.

Table II-2/1: Maximum group delay distortion with frequency

Frequency range (Hz)	Maximum group delay distortion (ms)
600 - 1000	4
1000 - 2600	2
2600 – 2800	6

4) Noise

The level of psofometric weighted noise at an NTP is less than -52 dBmp.

5) Variation of loss with input level

The transmission loss of a connection can be dependent on the level of the input signal. For the frequency range 700 to 1100 Hz, the variation of loss with signal input level, compared with an input level of -10 dBm, is within the mask of figure II-2/2.

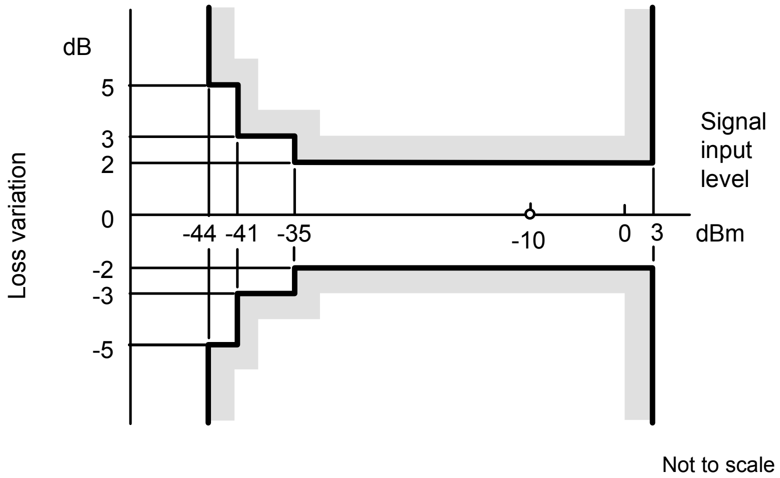


Figure II-2/2: Variation of loss with input level

6) Discrimination against out-of-band signals

With any sine-wave signal applied at one NTP of a connection, with a frequency above 4,6 kHz and a level of -25 dBm0, the level of any image frequency produced at the other NTP of the connection is at least 25 dB below the level of the test signal.

7) Total distortion, including quantizing distortion

For a sine-wave signal of 1020 Hz applied at one NTP of a connection, the signal to total distortion ratio (including quantizing distortion), measured at the other NTP of the connection, is above the limit of figure II-2/3.

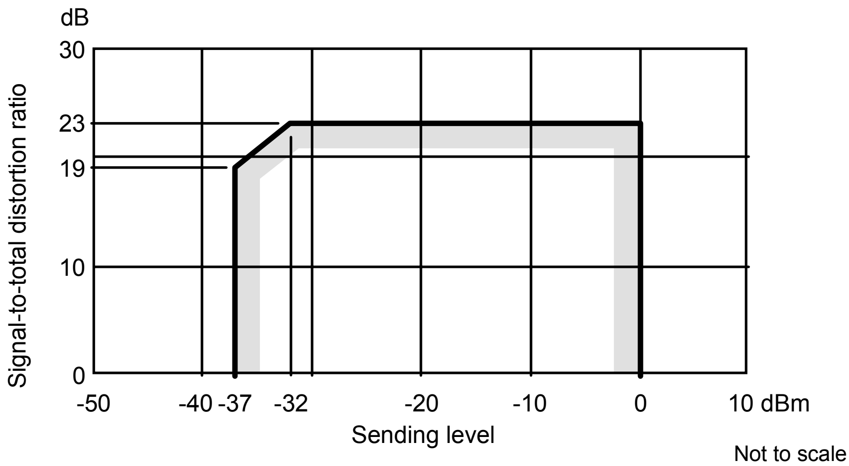


Figure II-2/3: Total distortion, including quantizing distortion

8) Impedance

The impedance of a telephone connection, seen from the NTP into the network, is very dependent on the situation; especially the length of the local loop has a great influence. Sometimes, the impedance of the connected TE at the other NTP has much influence.

In table II-2/2 a rough approximation of the impedance is given.

Because of the large variation in the telephone network, the measured impedance shall in many cases differ from the impedance provided in the table. The return loss, calculated with the formula below, will not be less than 6 dB. In many cases the return loss shall be more than 12 dB.

Table II-2/2: Mean values of the impedance at the NTP

Frequency (Hz)	Impedance Real part (ohm)	Impedance Imaginary part (ohm)
300	1100	-110
600	980	-260
1000	880	-310
1600	680	-290
2200	610	-250
2800	560	-220
3400	530	-200

Formula for the calculation of the return loss :

$$\textbf{Return loss} = 20 \times \log \frac{|Z_1 + Z_2|}{|Z_1 - Z_2|}$$

Z_1 = the actual impedance
Z_2 = impedance according to the table

9) Crosstalk

Crosstalk is the (unintended) penetration of signals from one telephone call into the connection of another call.
The telecommunication network is designed in such a way that, under normal circumstances and with connected TEs fulfilling the access requirements, audible crosstalk shall not occur. Due to failures in the network or in the TE, the defective equipment or lines may cause noticeable crosstalk to other connections.

10) Spurious signals

Spurious signals may appear on a telephone connection due to :
- Lightning or unintended connection with power lines or power circuits;
- Induction of signals from other circuits;
- Short interruptions.

Lightning into the ground can induce very high voltages in the local loop. These voltages will be conducted to both the exchange and the TE.
Induction of signals from other circuits (telecommunication and non telecommunication circuits) can occur if the symmetry to earth of the network or the TE is disturbed because of faults.

2.3.3 Transmission aspects of digital terminated connections

Connections in the fixed telecommunication network of KPN Telecom can at one or both sides be terminated at digital network terminations, i.e. ISDN network terminations.
The transmission properties as provided in section 2.3.2 for analogue terminated connections, are also valid for digital terminated connections, with the assumption that at the digital termination a (fictive) ideal coder/decoder is connected. The analogue side of the ideal coder/decoder is then to be seen as the analogue NTP. The statements about the impedance, as stated before for analogue terminated connections, are not applicable in this case.
The worst case values, as presented in section 2.3.2 for the analogue terminated connections, shall be much better for digital terminated connections, which results in a higher quality.

2.3.4 Data transmission

Data transmission via a telephone connection is possible when the data signal is, by means of a Voice Band Modem, converted into signals within the frequency band 300 to 3400 Hz of the telephone connection.
Most modems, which fulfil the ITU-T V-series Recommendations, will provide high performance voice band data connections.
The maximum achievable bit rate is determined by the quality of the connection and the working principle and quality of the modem.

In most cases, bit rates higher or much higher than 14,4 kbit/s are possible with high speed modems, e.g. V.34 type of modems. If one side of the telephone connection is digital, what is often the case for Internet access, very high bit rates up to 56 kbit/s with V.90/V.92 modems are possible.

3 PSTN call handling procedures

3.1 Introduction

In this chapter, a functional description is given about the behaviour of the PSTN with respect to the establishing, holding and releasing of a telephone connection, as this is perceivable on the subscriber line of the customer. This behaviour is called "call handling"; it is perceivable by way of signals and acoustic indications from the network to the line and whether or not the communication path between calling and called user is through connected in the network for end-to-end communication.
The description in this chapter is restricted to the basic call handling; procedures related to facilities (e.g. Direct Dialling In) and supplementary services (e.g. Calling Line Identification Presentation, Call Waiting Hookflash, Completion of Calls to Busy Subscriber) are dealt with in the detailed signalling descriptions.

The call handling process is described in a functional way; the signals and acoustic indications are defined in the actual meaning for the user without notice of the physical appearance on the line. In doing this, the description of the basic call handling process is general and not dependent of different physical behaviour related to different exchanges and subscriber line signalling systems. The physical conditions on a subscriber line at the network termination point will be described comprehensively in following chapters.

For one and the same communication connection, there is a distinction to be made between the call handling process at the calling user side (the call originating side) and the call handling process at the called user side (the call terminating side). These originating and terminating call processes are described separately.
When in the following an action from a calling or called user is mentioned, this shall be interpreted as an action from the Terminal Equipment (TE) of the calling or called user, irrespective whether the action is initiated by the user or by an automatic function in the TE.

The description of the call handling procedures does contain the call handling related timing information, e.g. the time during which a tone is sent or ringing voltage is supplied to the network termination.

Note:
In this publication the terms "outgoing" and "incoming" are avoided as much as possible, because the meaning of such term for a user or a TE is opposite to the meaning for the network. Therefore, a consequent use is made of the term "originating call" for a call "outgoing" from a user's TE and "terminating call" for a call "incoming" to the user's TE.
Many times the abbreviation IS for (telecommunication) infrastructure is used in the meaning of the PSTN or the local exchange of the telecommunication network of KPN Telecom.

3.2 Functional conditions of the subscriber line

The following conditions of the subscriber line can be identified:
- idle;
- occupied originating;
- occupied terminating;
- locked out;
- blocked.

The condition "idle" is the normal condition of the subscriber line when the line is not in use for a call and is available for a new originating or terminating call. The idle condition is always present as signalling state in signalling systems.

The conditions "occupied originating" and "occupied terminating" are present as long as the line is involved in an originating or a terminating call respectively. During one or the other condition, the originating respectively the terminating call process is active. In such process, a number of call handling phases will be passed through. The originating and terminating call processes are described in the following sections 3.3 and 3.4 respectively.

During the condition "locked out", there is no call handling process active any more, but the line is not available for a new call.
The condition appears when the network detects an unjustly occupation of the line, i.e. the line is seized lengthy for an originating call and no dialling information is received, or the network has indicated that, at the end of a call, the connection in the network has been released, but no release indication has been received in time from the user's TE.
As soon as in this condition a release indication is received from the TE, the "locked out" condition will, eventually after some delay time, cease and the line becomes "idle". The delay time may last from some seconds to several minutes.
The "locked out" condition may also appear when the exchange detects a line failure in the subscriber line.
The network will make sure that, during the "locked out" condition of the line, there will be no needless occupation of network resources; also the provision of the DC line feeding may be limited, i.e. the DC loop current may be restricted.

By means of the condition "blocked", the network can indicate that call handling for a line is not possible, i.e. the line is taken out of service. This may occur because of technical or administrative reasons, or a (temporary) line failure has been detected in the exchange, or during maintenance actions (e.g. testing of lines, repair of a broken cable). On analogue subscriber lines, the "blocked" condition can be noticed by the absence of DC line feeding voltage.

In figure II-3/1 the mutual relations of the above described conditions are presented.

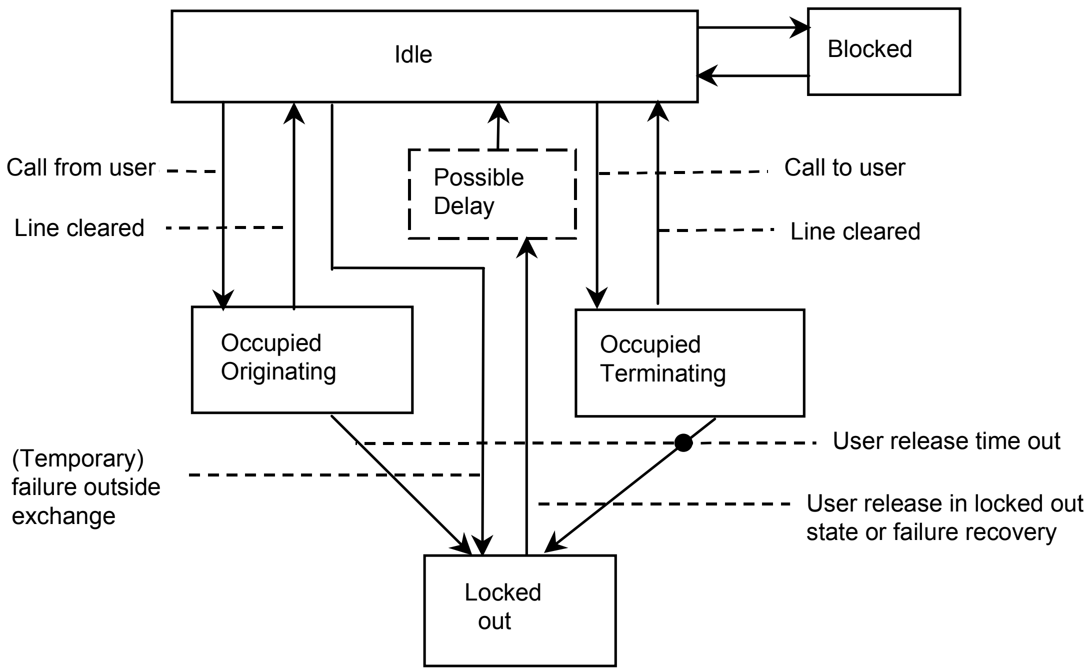


Figure II-3/1 Functional conditions of the subscriber line

3.3 Originating call process

3.3.1 General

An originating call handling process is started in the PSTN exchange as soon as a subscriber line in "idle" condition is seized by a TE.
As a normal procedure of the originating call handling process, the following functional phases are passed through:

- seizing phase;
- dialling phase;
- indication phase;
- communication phase (originated from TE);
- release phase, forward (from originating TE to network);
- release phase, backward (from network to originating TE).

Each of these phases can consist of one or more signalling states and related physical appearance on the line. This is dependent of the signalling system as in use on the line.

In case of a not successful call set-up (e.g. premature release, congestion in the network, called party busy) or a not answered call, one or more of the phases of the normal procedure will be left out.

The above mentioned phases are described below in more detail; in figure II-3/2, the originating call process is visualized.

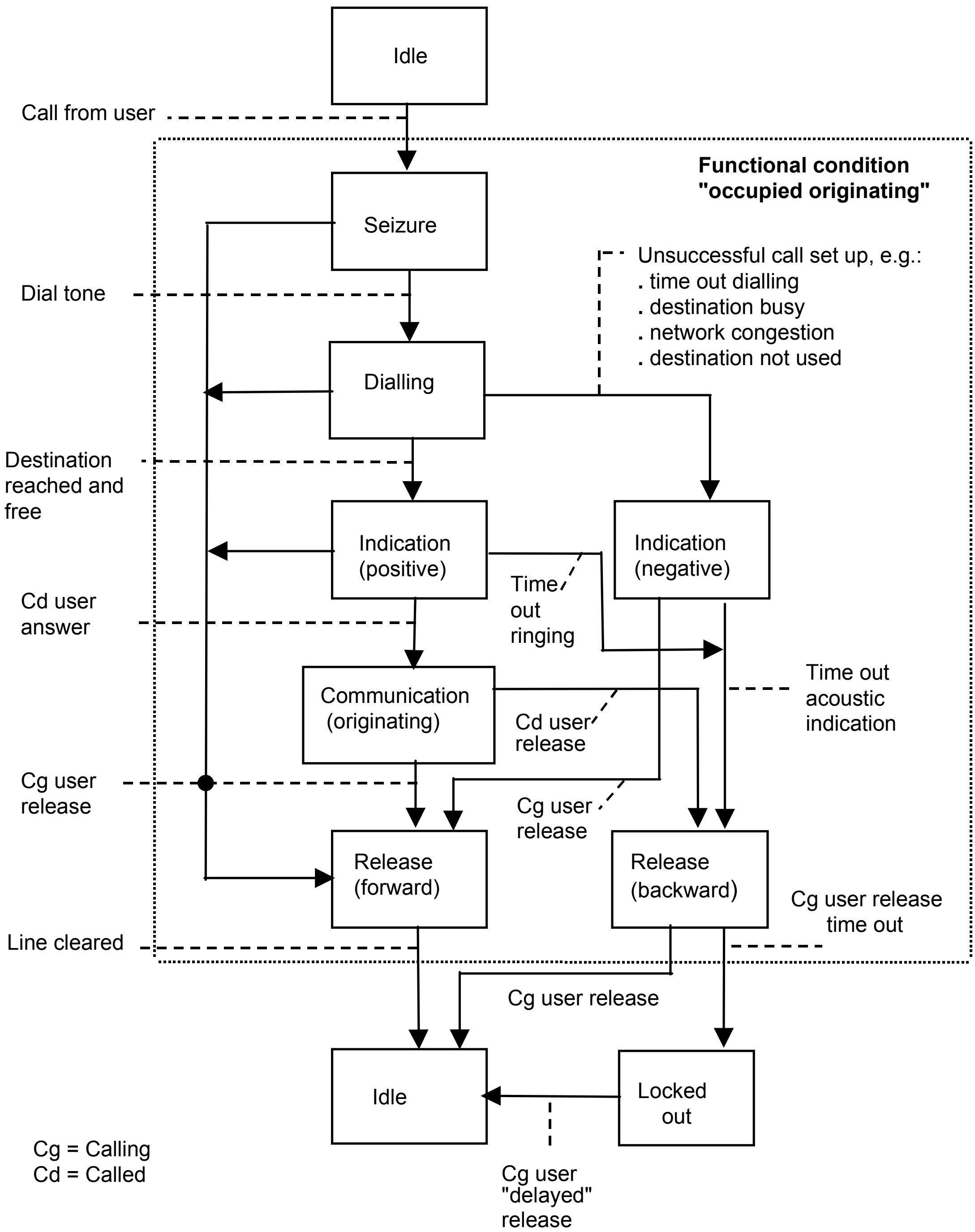


Figure II-3/2 Originating Call process

3.3.2 Seizing phase

The seizing of the line for an originating call from the user is effectuated by sending the signal "seizing-TE" from the TE. With this signal, sent in the idle line condition, the TE indicates the wish for originating a call.

Normal procedure
As soon as the "seizing-TE" signal is detected in the PSTN (further also indicated as infrastructure or IS), the originating call process for this line will be started. The exchange has first to search for a free receiver for

dialling information, will connect the subscriber line with this receiver and then will send "dial tone" to the line. The "dial tone" is the indication to the user or the TE that the seizing phase has ended and the dialling phase has been entered. Only from that moment on, it is assured that dialled digits will be received correctly.
The time between sending of "seizing-TE" and reception of "dial tone", the "dial tone delay", is dependent of the call attempt intensity and the traffic load of the local exchange; it may be assumed that this delay will be shorter than 2 s (except for very exceptional situations).

Abnormal procedure
In exceptional cases with very high traffic demands in the local exchange, the "dial tone delay" may last several seconds; in extreme situations it may be possible that the "dial tone" can not be provided at all. For automatic calling TEs, a dial tone detector is not mandatory; according to ETSI TBR 21 a time out for starting dialling after seizing, with a minimum of 2,7 s, is allowed. It may be clear that in those exceptional cases the first digit(s) may not be detected and the call attempt may fail, i.e. the exchange will have received an incomplete number or just a local subscriber number in stead of a national telephone number.
"Call collision" on both-way lines is another reason of failing to start the originating call process; at about the same time that the TE sends the "seizing-TE" signal for a originating call, the exchange has to offer a terminating call to the line. In such case, the terminating call has precedence. Dependent on the signalling system, the "seizing-TE" signal will then be interpreted as the "answer-TE" signal of the terminating call process, or TEs (i.e. PBXs) have to switch from the originating call process to the terminating call process within a defined time; see the chapters with the signalling system descriptions.
When the calling user or the TE releases already in the seizing phase, the forward release phase is entered.

3.3.3 Dialling phase

During the dialling phase the user or the TE has to send to the network the address of the wanted destination. For a normal call this address is the telephone/ISDN number of the destination according to the "dialling plan"; this means that in the dialled number, prefixes and/or escape codes are included as defined in the dialling procedures.
In case the call is originated for the control of a supplementary service, the control command is given in this phase, e.g. activation of call forwarding unconditional to a destination with telephone number TFN by sending the digit string: ∗21∗<TFN>#.

The dialling process is supervised by time guards:
- the "dial tone send guard" with a duration of 20 - 40 s for the first digit;
- the "digit entering guard" with a duration of 6 - 8 s for the following digits.

The DTMF codes '∗' and '#' are to be seen as digits.
As soon as the first digit is received, the sending of "dial tone" is stopped.

Successful call procedure
The dialling phase for a successful call ends when the network has determined that:
- a complete and allocated number has been received;

- the destination network termination is reached;
- the destination network termination is free;
- the seizure/ringing phase of the terminating call process at the destination network termination is entered.

The network will send the "number received" signal. This signal indicates that the indication phase is entered; in case of a successful call a positive indication, i.e. "ringing tone", is provided. In case the call is answered very quickly at the terminating side, e.g. in case of call collision at a terminating both-way line, the "ringing tone" may be provided very short or even not at all.
The time between the sending of the last digit of the destination number by the user or the TE and receiving the "number received" signal and hearing the "ringing tone" is called "post dialling delay". With the fast signalling system as in use in the network of KPN Telecom, the "post dialling delay" for national connections shall be 1 to 2 s at the utmost. For international connections, this delay may be longer and may last for more than 5 s; some networks may provide during this delay time a special tone, which is called network "routing tone".
In case that a control command for a supplementary service is successfully executed, the local exchange will also send the "number received" signal and enter the indication phase with a positive indication.

Unsuccessful call procedure
One cause of unsuccessful dialling is the premature release of the calling user or the TE during dialling; the dialling phase is then left and the forward release phase is entered.
A lot of other causes for unsuccessful call set up can be identified, e.g.:

- not receiving all digits of the destination number in time (incomplete number);
- destination not existing (unallocated number);
- destination busy;
- network congestion;
- not allowed destination;
- not valid control command for supplementary service.

In these cases, also the dialling phase is ended and the indication phase entered by sending the "number received" signal; a negative indication for unsuccessful call set-up is given.

3.3.4 Indication phase

During the indication phase (the "number received" signal has been sent from the IS) the network provides information about the result of the call establishment by means of an acoustic indication.

Positive indications
For a successful call to another user, the positive indication is the "ringing tone", which means that the call is successful routed through the network and the seizure/ringing phase at the terminating side is entered. In this phase of the originating call process, the communication path is not yet through connected in the network. The "ringing tone" is provided during a limited time, the ringing time guard; this is the time during which the call is offered by supplying "ringing signal" at the terminating side. For national calls, this is between 60 and 100 s; for international calls, this time may last between 2 and 4 minutes according to ITU-T Recommendations.

The phase with positive indication ends when:

- the called user or TE answers the call: the communication phase is entered;
- the calling user releases the call: the release forward phase is entered;
- the ringing time elapses: the release backward phase is entered.

In case of a control command for a supplementary service, the positive indication is the "acceptance tone" which has the same characteristic as the "dial tone"; the "acceptance tone" is given for about 5 s. At calling user release, the release forward phase is entered; at time out, the release backward phase is entered.

Negative indications

With a negative indication, in the form of a tone or an announcement, the cause of an unsuccessful call is given. A tone is provided during 10 - 20 s; an announcement is provided so long, that the announcement can be listened to at least two times. In case of a failed control command for a supplementary service, the "rejection tone" is given for about 5 s.
In table II-3/1 the relations between the most important causes of call failures and provided indications are presented.

Table II-3/1 Relations between causes of unsuccessful calls and provided indications

Cause	**Acoustic indication**
Destination busy	Busy tone
Network congestion	Congestion tone
Destination not existing (e.g. unallocated number, blocked for administrative reasons); the directory service can provide further information	Information tone
Not receiving digits in time	Congestion tone
Number has been changed	Specific announcement or information tone
Temporary technical failure	Congestion tone
Failed control command for supplementary service	Rejection tone (same characteristic as congestion tone)

From the foregoing it can be noted that, when "congestion tone" is received, this not always means that there is network congestion. In all call failure cases, for which no specific indication exists, the "congestion tone" is provided; the "release tone" in the release phase is also provided with the same characteristics as the "congestion tone".

The indication phase ends and the release phase is entered:

- when the calling user releases: the release forward phase is entered;
- when the listening time elapses: the release backward phase is entered.

3.3.5 Communication phase

The communication phase is entered when at the terminating side the call is answered; the sending of "ringing tone" is stopped and the communication path in the network is through connected. The charging for

the call is started. It is possible that for international connections, where older types of signalling systems are still in use in foreign networks, there is some delay in through connection.
The entering of the communication phase is not indicated to the calling user; no answer signal is given.
The communication phase is ended and the release phase entered when one of the parties releases the call.

3.3.6 Release phase

The procedure in the release phase depends on the party which releases first:
- if this is the calling party: the release forward procedure follows;
- if this is the called party: the release backward procedure follows.

Release forward
Forward release can be initiated by the calling party in all phases of the call. The TE sends the "release-TE" signal to the exchange. On receipt of this signal, the exchange terminates the call and, with the sending of the "release-IS" signal, the line to the calling user is set in the idle condition. The "release-IS" signal is only applicable from the indication phase on, i.e. after the "number received" signal is sent by the exchange.

Release backward
When the called party has released (and the calling party hasn't yet), the connection in the network will be released and the release backward phase is entered. The release backward phase is also entered when in the indication phase, the calling user has not released before the time guard for listening to the acoustic indication elapses. In the release backward phase, the exchange will during some time (10 - 20 s) send the "release-IS" signal and at the same time provide "release tone" to the originating line. This "release tone" has the same characteristics as the "congestion tone".
As soon as the calling party releases by sending the "release-TE" signal, the line is set in the idle condition.
When the calling party does not release within above mentioned time, the line will be set in the locked out condition; see section 3.2.

3.4 Terminating call process

3.4.1 General

A terminating call handling process is started in the PSTN exchange as soon as a call has to be offered to a line in "idle" condition.

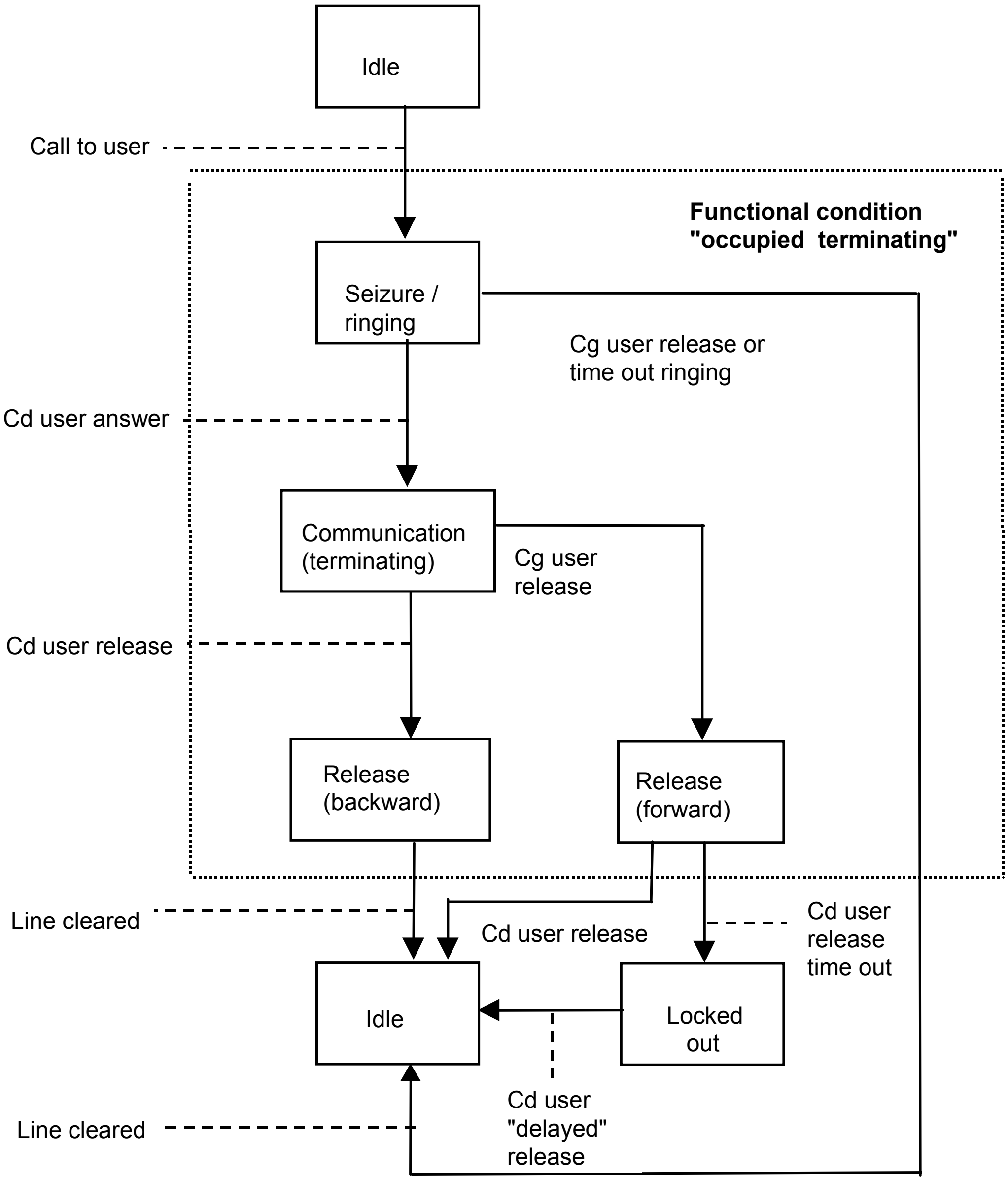
Note:
Line hunting for a free (= idle) line of an Multi-Line Termination and procedures related to facilities (e.g. Direct Dialling In) and supplementary services (e.g. CLIP) are not described in this section.

As a normal procedure of the terminating call handling process, the following functional phases are passed through:
- seizing/ringing phase;
- communication phase (terminated to TE);
- release phase, forward (from network to terminating TE);

- release phase, backward (from terminating TE to network).

Each of these phases can consist of one or more signalling states and related physical appearance on the line. This is dependent of the signalling system as in use on the line.

The above mentioned phases will be described in more detail; in figure II-3/3, the terminating call process is visualized.



Cg = Calling
Cd = Called

Figure II-3/3 Terminating call process

3.4.2 Seizing/ringing phase

The seizing of the line for a terminating call is effectuated by sending both the signals "seizing-IS" and "ringing signal" from the exchange. With these signals, sent in the idle line condition, the network indicates the offering of a terminating call to the line.
At the same time, i.e. during the seizing/ringing phase, "ringing tone" is sent to the calling party; see the indication phase of the originating call process.
The communication path between calling and called party is not through connected yet. The call progress is guarded by the ringing time guard with a duration between 60 and 100 s.
The "ringing voltage signal", with a time duration of about 1 s, is cyclic repeated; the pause duration is about 4 s. The pause between the 1st and the 2nd "ringing voltage signal" may be shorter.

Normal procedure
The terminating call can be accepted by sending the signal "answer-TE" to the network. The network will then as quickly as possible stop the sending of "ringing signal" to the called party and "ringing tone" to the calling party, and through connect the communication path between them. The communication phase is then entered.
It is not possible for the called party to release the call without having answered the call.

Abnormal procedure
If the calling user releases before "answer-TE" is received or the ringing timer elapses, the terminating call process is stopped by the network by sending the signal "release-IS". With this signal, the line comes in the idle condition again or, if the TE had come in a special seizing condition, the TE shall restore the idle condition immediately.

3.4.3 Communication phase

The communication phase is entered when the "answer-TE" signal has been received.
For international connections, where older types of signalling systems are still in use, there may be some delay in through connection of the communication path.
Because there is no answer signal available at the originating call side, it is advisable for automatic answering TE to send, after the "answer-TE" signal, an in-band answer indication to the calling party, e.g. a spoken answer message, or a 2100 Hz answering tone as defined for modems or fax apparatus. It should be avoided that in this situation a tone is sent, which could be interpreted by the calling user as a network indication tone as mentioned in the originating call process.

The communication phase is ended and the release phase entered when one of the parties releases the call.

3.4.4 Release phase

The procedure in the release phase depends on the party which releases first:

- if this is the calling party: the release forward procedure follows;
- if this is the called party: the release backward procedure follows.

Release forward

When the calling party has released (and the called party hasn't yet), the connection in the network will be released and the release forward phase is entered. In this phase, the exchange will during some time (10 - 20 s) send the "release-IS" signal and at the same time provide "release tone" to the called party's line. This "release tone" has the same characteristics as the "congestion tone".
As soon as the called party releases by sending the "release-TE" signal, the line is set in the idle condition.
When the called party does not release within above mentioned time, the line will be set in the locked out condition; see section 3.2.

Release backward

Backward release is initiated by the called party. The TE sends the "release-TE" signal to the exchange. On receipt of this signal, the exchange terminates the call and, with the sending of the "release-IS" signal, the line to the called user is set in the idle condition.

4 PSTN network terminations

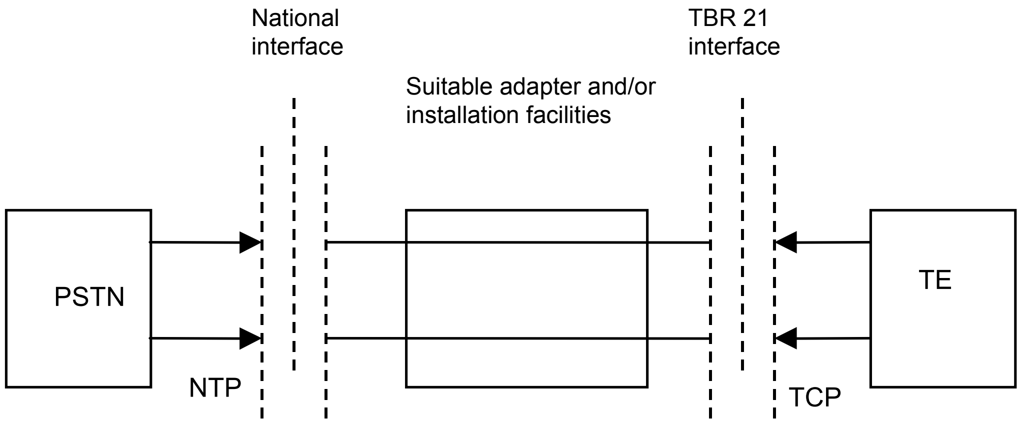
4.1 Introduction

The PSTN Network Termination (NT) can be defined as:
All physical connections and their technical access specifications which form part of the Public Switched Telephone Network and are necessary for access to and efficient communication through that PSTN.

In ETSI the following terms are defined for identifying the point of connection of Terminal Equipment (TE) with the PSTN:

- Network Termination Point (NTP): The physical point at the boundary of the PSTN intended to accept the connection of a Terminal Equipment (TE);
- Terminal Connection Point (TCP): The point of the TE intended to be connected to the PSTN, either directly or indirectly via a suitable adapter, installation facilities or another serially connected TE.

Figure II-4/1a, as derived from ETSI TBR 21 and ETSI EG 201 120, illustrates such.



Source: ETSI: TBR 21 / EG 201 120

Figure II-4/1a Network Termination Point and Terminal Connection Point

In other words, the NTP is the point at which a TE with its Terminal Connection Point (TCP) can be connected.
In general, customers will have in house cabling and provisioning to connect via such cabling more than one TE to the PSTN NTP. This customer's premises installation is the customer's own responsibility. Further information about these installation matters, is provided in following section and in chapter 5.

In this chapter the analogue NTs, as provided by the PSTN of KPN Telecom for narrowband services, are described.
Because of the different characteristics, a distinction is made between:

- Single-Line Termination (SLT);
- Multi-Line Termination (MLT).

These are dealt with in respectively sections 4.3 and 4.4.

The relation between the call handling process and the physical phenomenon, as to be perceived on the line, is called signalling. A general introduction to the analogue subscriber line signalling is given in section 4.5.

Beside the general information about the analogue NTs and the subscriber line signalling, detailed information is provided about the electrical conditions at the NTPs, irrespective whether the NT is an SLT or an MLT; this information is contained in section 4.6.

4.2 Physical presentation of the PSTN network access

Already a long time ago the well known wall mounted 4-terminal telephone socket is applied as the Network Connection Point (NCP) in the PSTN of KPN Telecom; it represents a means for simple and flexible connection of TE, which is provided with a corresponding plug. In practice this plug/socket system is also in use in the customer's premises wiring.

As a consequence of the liberalization of the TE market, KPN Telecom has defined the IS/RA (separation) point as the means to make a clear distinction between the network infrastructure (IS) and the customer's premises installation with Terminal Equipment (Dutch: Randapparatuur RA) and in house cabling with sockets for plugging in the Terminal Equipment.

The physical presentation at the IS/RA point is a connection box in which the infrastructure cable wires are fitted and are through connected to a customer's accessible part of the box, the Network Connection Point (NCP).

As for the original analogue telephone service, no line terminating equipment is needed, the NCP is also the Network Termination Point (NTP).
With the introduction of ADSL in combination with PSTN, a filter unit (POTS splitter) may be located between NCP and NTP.

Figure II-4/1b illustrates such.

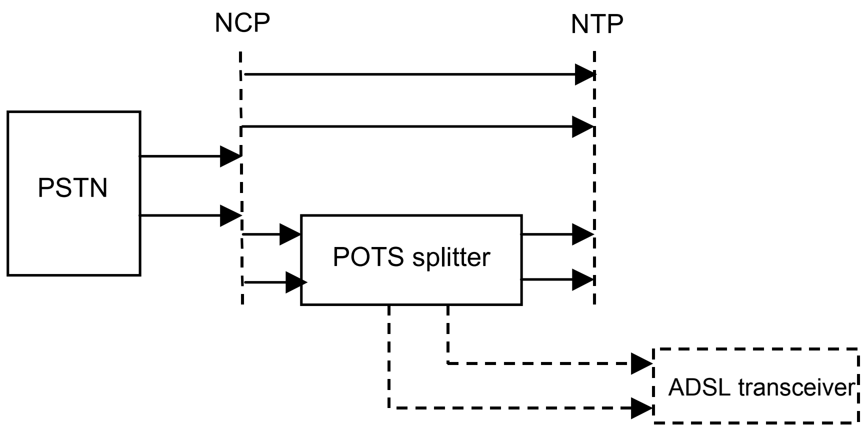


Figure II-4/1b Network Connection Point and Network Termination Point

Detailed information about the physical presentation of the 2-wire network access and the IS/RA connection principle is provided in Part V of the present publication. Information on the interworking with ADSL is provided in Part IV-A of this publication.

4.3 PSTN single-line termination

A Single-Line Termination (SLT) is the physical connection and access capability of a customer's NTP with the PSTN infrastructure, which consists of one 2-wire line for one speech connection.
The distance between the customer's NTP and the line interface in the local exchange can be expressed as the value of the line resistance. The value of the resistance of the subscriber line in the network of KPN Telecom is between 0 and 1000 Ω; dependent of the diameter of the wires, the maximum resistance corresponds to a length of the line between 5 and 8 km.
The line of an SLT is always a both-way line, which means that both originating and terminating calls are applicable for that line.

A PSTN Single-Line Termination (SLT) is always provided with a single telephone/ISDN number, the subscriber number. This number identifies the customer's NTP:

- In case of originating calls, the identity is used for e.g. billing and calling line identification purposes;
- In case of terminating calls, the identity is used for routing and offering calls to the SLT of the customer.

If a customer has more than one subscriber line, e.g. x lines, but each of these lines is identified by its own dedicated telephone number, then that customer is provided with x individual SLTs.

On an SLT, only one telephone conversation at the same time is possible. If for the SLT the Call Waiting Hookflash supplementary service is applicable, the customer can at the same time be involved in two telephone calls, one in speech condition and the other in waiting or hold condition.

In general the mean traffic load on an SLT will be low; not all subscribers are willing to make telephone calls at the same time and the average holding time of calls will be restricted. It is not necessary that communication paths through the exchange are available for all connected SLTs at the same time. Therefore, the first switching stage of the exchange provides a concentration of the connected SLTs to a smaller number of trunk lines. The reduction factor is related to the demands in the busy hour for that exchange. The foregoing means that a customer, when initiating a call, has a chance to meet congestion because of unavailability of network resources in the local exchange. The increasing use of Internet tends to increasing values for the mean holding time of calls.

It is general practice to indicate the two wires of an SLT as the a- and b-wire. The a- and b-wire of an SLT is one twisted pair out of a cable with a lot of twisted pairs. The impedances of the a-wire to earth and the b-wire to earth shall be nearly equal in order to avoid cross talk to the other lines in the cable; the a-wire and b-wire shall be symmetric with respect to earth.

The PSTN exchange provides the DC line feeding voltage to the line. In idle condition of the line, the polarity of the feeding voltage is not fixed; the a-wire can be positive or negative with respect to the b-wire. TE, intended to be connected to an SLT for the basic telephone service, shall therefore operate independent of the polarity of the feeding voltage. TE with CLIP function, shall use the provided reversal of the idle polarity for initiating the CLIP procedure; see the related section in chapter 6.

The electric conditions at the SLT are provided in section 4.6; the detailed signalling procedures on SLTs are described in chapter 6.

4.4 PSTN multi-line termination

A Multi-Line Termination (MLT) is the physical connection and access capability of a customer's NTP with the PSTN infrastructure, which consists of a group of 2-wire lines with a minimum of two lines; the number of possible independent speech connections at the same time is equal to the number of lines.
The distance between the customer's NTP and the line interfaces in the local exchange can be expressed as the value of the line resistance. The value of the resistance of the subscriber lines in the network of KPN Telecom is between 0 and 1000 Ω; dependent of the diameter of the wires, the maximum resistance corresponds to a length of the line between 5 and 8 km.

An MLT can be used to connect individual "simple" TE (TE as applicable for SLTs) to each line, or to connect a Private Branch Exchange (PBX) or private network, or to connect other more or less "complex" TE.
The traffic load on MLTs can be diverse: for some MLTs the traffic load per line may be as low as the traffic load which can be expected on an SLT; for other MLTs it may be much higher than on SLTs.

As part of the subscription, some basic features have to be attached to the MLT in order to define the termination identity and the traffic handling capability of that termination. These are part of the attachment agreement with the customer.
In the following, short descriptions of these features are given.

Traffic handling
The lines of an MLT can, in agreement with the customer, be split up in different line groups, i.e. line groups for originating-only traffic, terminating-only traffic and both-way traffic.

Line Hunting (LH)
The Line Hunting (LH) feature is always present and active in case of an MLT.
As far as the PSTN is concerned, LH is applicable for the line groups for terminating traffic. It is advisable that a connected PBX is also provided with a LH function for its originating traffic, and further that the hunting methods for the both-way line group in the PSTN exchange and the PBX are chosen such that traffic handling on this line group is done as efficient as possible.

The function of LH in the exchange is to search for a free line in the line groups for terminating-only and/or both-way traffic for offering a call to the customer's installation.

Per line group, one of the following hunting methods can be provided:
a) Sequential hunting with fixed starting point (also called "home start"): the lines are arranged in a chain with a first and a last member; the search for a free line is always started at the first line and continued in the fixed order until a free line is found or the last member is reached.
b) Cyclic hunting with semi-random or random starting point: the lines are arranged in a circular chain; the search is started at a random chosen line, or at the line following the line on which the last terminating call was offered; the search is continued in a fixed order until a free line is found or the starting point is reached again.

With the cyclic method, traffic handling may have little trouble with temporary line failures and all lines are used equally. On a both-way line group, the sequential method may be advantageous because, when the ordering of the 'interface chain' in PSTN and PBX are opposite, the risk of call collision will be low.

Group Number (GN)
The Group Number (GN) feature makes it possible that a customer with a Multi-Line Termination (MLT) can be reached via more than one line with one telephone number, the group number. The GN identifies the network termination in the same way as the subscriber number does for an SLT. The Customer's Premises Equipment (CPE) can at the same time be involved in more than one telephone conversation, i.e. one telephone conversation per line.
As an option for an MLT with GN, one or some lines can be made direct accessible with its own telephone number; such number(s) is(are) then called "sub-number(s)" of the GN. If a call is received for such "sub-number" and the related line is busy, the call will be released with the indication busy; line hunting is not applicable for such "direct access" calls. The application of such "direct access" lines can be for offering calls directly to special TEs, e.g. non-voice TEs like fax apparatus. These direct access lines remain part of the line group for terminating or both-way traffic; for calls to the group number, they remain part of the to that line group related line hunting procedure.

Direct Dialling In (DDI)
The Direct Dialling In (DDI) feature makes it possible for a customer with an MLT, that the individual TEs, connected to the customer's PBX or private network, can directly be reached with a "normal" telephone number, the DDI number. For this aim, a range of consecutive numbers is allocated to the MLT; i.e. a part of the public telephone numbering plan is allocated as internal numbering plan of such PBX or private network. For originating calls, the identity of the MLT with DDI, e.g. for identification services, is one of the DDI numbers which is nominated as the "default DDI number". The Customer's Premises Equipment (CPE) can at the same time be involved in one telephone conversation per line.

Other facilities
On an MLT, facilities like Malicious Call IDentification (MCID) and outgoing call barring can be provided.

On small sized MLTs without DDI, i.e. an MLT with maximum 6 lines with the Group Number (GN) feature, also Call Forwarding Unconditional (CFU) can be provided; relevant TE is required to cope with the related control command (e.g. ∗21∗Forwarded To Number#) and with the special dial tone, which is provided when CFU is activated.

It is general practice to indicate the two wires of an SLT as the a- and b-wire. The a- and b-wire of an SLT is one twisted pair out of a cable with a lot of twisted pairs. The impedances of the a-wire to earth and the b-wire to earth shall be nearly equal in order to avoid cross talk to the other lines in the cable; the a-wire and b-wire shall be symmetric with respect to earth.

The line feeding voltage for the lines in an MLT is provided by the PSTN exchange; the voltage is supplied such that in idle condition of the line the a-wire is positive with respect to the b-wire.
For the call handling related signalling from the exchange to the TE, the polarity of the feeding voltage is used, i.e. polarity reversals are applied.
In case of an MLT with GN, TE with this polarity dependent signalling can be connected.Polarity independent TE, as for SLTs, can also interwork with this type of network termination.
In case of an MLT with DDI, only TE with the polarity dependent signalling is to be connected.

The electric conditions at the MLT are provided in section 4.6; the detailed signalling procedures on MLTs are described in chapter 7.

4.5 Subscriber line signalling, general aspects

Signalling on the subscriber line is to be understood as all the information exchange between a TE and the PSTN for the purpose of establishing, holding and releasing of telephone calls and for the support of features and facilities.
For this aim, a lot of signals with different physical properties and meanings are used on the analogue subscriber lines. Which signals are applied on a subscriber line and the procedures of use, is defined in the applicable signalling system. The applicable signalling procedures are dependent on the type of network termination and the applicable features and/or facilities; see the chapters 6 and 7 for single-line and multi-line signalling respectively. Common is, that the signalling procedures are always based on and derived from the call handling procedures as described in chapter 3.

Note:
In section 2.2.1 information is provided on the application by modems and fax apparatus of the 2100 Hz answering tone, as defined in ITU-T Rec. V.25. This is an end-to-end signalling procedure and will not be described as part of the subscriber line signalling.

Related to the physical appearance on the analogue line, the signals can be categorized as follows:
- Continuous DC signals;
- Pulsed DC signals;
- AC signals;
- Tone code signals;

- Audio signals.

A short explanation of each category is given in the following; the electric conditions on the line are described in detail in section 4.6.

Continuous DC signals

The continuous DC signals represent the line states or signalling states which, in general, correspond to the phases in the call handling process. There is no time duration specified as part of the signal; the duration of the signalling state is determined by the call handling process.
The physical appearance is determined by applying between the a- and b-wire of the line, a DC feeding voltage in the exchange and a load resistance in the TE. In this way the subscriber line constitutes a "DC loop". For the exchange, the value of the DC loop current is a measure for the load resistance in the TE; for the TE, the direction of the DC loop current (or the absence of loop current) is a measure for the polarity of the feeding voltage (or the absence of feeding voltage) in the exchange.
Changing from one loop condition to another initiates a transition from one signalling state or call phase to another. For reliable interworking, it is necessary that an initiated change in loop condition at one end of the line is present during some time before it is recognized at the other end as a state transition; in the signalling systems, a minimum and a maximum recognition time for a signal is defined. In some states, where a distinction has to be made between a change to another state or the presence of a pulsed DC signal, the recognition time may be long (longer than the maximum duration of the pulsed signal).
Examples of continuous DC signals on the subscriber line are: seizing, ready to receive, number received, answered, release.

Pulsed DC signals

Pulsed DC signals are timed changes of the DC loop condition. Also for this category of signals, a minimum and maximum recognition time is defined. The maximum recognition time is shorter than the minimum recognition time for the related continuous DC signal, which would initiate a change of signalling state.
Examples of pulsed DC signals are: dialling pulses (decadic dialling), hookflash signal, MCID pulse.

AC signals

AC signals are defined here as the low frequency signals below the voice band (300 - 3400 Hz), i.e. the 25 Hz ringing signals.

Tone code signals

Tone code signals are the Dual Tone Multi Frequency (DTMF) signals in the 3,1 kHz voice band as applicable for DTMF dialling.
DTMF dialling is the international standardized system for sending the called party number to the PSTN for establishing a telephone connection, or the digits for the control of a supplementary service. It is also used in the terminating call process of the PSTN of KPN Telecom for delivering the called DDI number to a PBX with the DDI feature and for delivering the calling party number to the called party's network termination as part of the CLIP supplementary service.
The DTMF signalling system is also used in end-to-end communication of users with e.g. voice response systems; this application is outside the scope of this document.

The original CEPT standards on the DTMF system are now succeeded by the ETSI Standards ES 201 235, parts 1 to 4.

Audio signals

Audio signals are the tones and announcements which are applied by the PSTN in the 3,1 kHz voice band as positive and negative indications in the call handling process. The aim is to provide the user of a voice telephony TE with audible information on the call progress. For non voice TE and for automatic calling and answering TE, it may be necessary or desirable to recognize the tones as provided by the network as part of the call handling process.

Examples of the network tones are: dial tone, special dial tone, ringing tone, busy tone, congestion tone, information tone, release tone. The applicability of these tones is described in the call handling procedures in chapter 3; when "dial tone" is mentioned there, it will be the "special dial tone" in the case that the CFU supplementary service is activated.
In case of the CWH supplementary service, a new incoming call will be announced to the NTP with a call waiting tone while the first call is still active in the communication state.

Unfortunately, there are a lot of deviations in the characteristics of the tones in the different PSTNs in the world. ITU-T Rec. E.180 provides recommendations with respect to the PSTN tones and Supplement No. 2 to the ITU-T E-series Recommendations contains information on the characteristics of the applicable tones in different countries. Information on the European situation can also be found in EN 300 001 from ETSI.

The tones, as applied in the PSTN of KPN Telecom, conform to Recommendation E.180; see the characteristics as described in section 4.6.6 hereafter.
In case of international calls, a user can be informed with a tone which is provided by a foreign network; e.g. the ringing tone is normally sent by the destination exchange in the foreign network.
In some countries a "routing tone" is applied during the time that the network is at work on establishing the connection. For example in France, a routing tone is applied with a frequency of 425 Hz and a cadence of 50 ms on and 50 ms off; this tone should not be interpreted as e.g. congestion tone.

4.6 Electric conditions of supplied network signals

In this section the principle of supplying the network signals to the analogue subscriber line is described and the electric conditions of the signals are provided.
Further, some attention is paid to the phenomenon of external disturbing signals on the line.

4.6.1 Principles of supplying signals to the line

During the establishment phase, communication phase and release phase of a call, the exchange has to supply signals to the Network Termination Point (NTP), where TE is or can be connected. The schematic principle of the signal supply is presented in figure II-4/2.

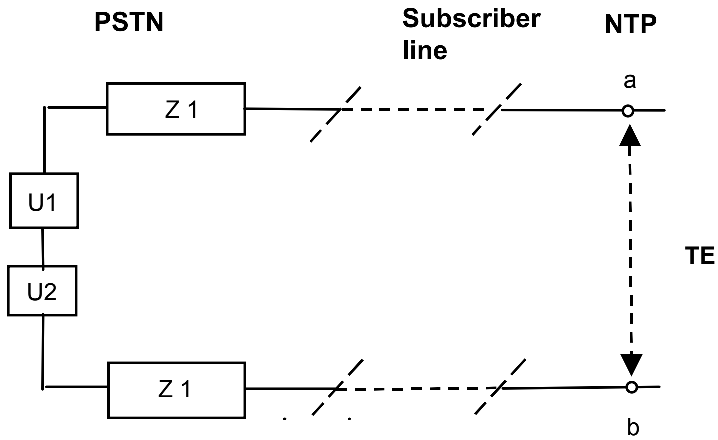


Figure II-4/2 Schematic principle signal supply

The boxes in the figure have the following meanings:

U1 = DC voltage supply for DC loop signalling (including decadic dialling) and feeding of "simple" TE.

U2 = AC voltage supply for tone code signals, tones and announcements, ringing voltage.

Z1 = Internal resistance of the supplies.

4.6.2 Continuous DC signals / line feeding

The PSTN exchange supplies a DC voltage between the a- and b-wire of an analogue subscriber line.
The value of the voltage at the NTP is dependent of a lot of conditions, e.g. the type of exchange, the resistance of the line, the loop termination in the connected TE, whether the mains voltage is present in the exchange or the exchange has to rely on emergency powering.

For a line in service, i.e. a line in idle, occupied and locked out condition, the DC voltage is always present at the NTP.
A line may be taken out of service (set in the blocked condition) by the exchange, which can be noticed by the absence of the DC feeding voltage.

In the in-service condition of the line there may be short voltage interruptions due to actions in the exchange, e.g. changing from one feeding bridge to another, reversing voltage polarity, testing of the line circuitry, and such; the interruptions are kept as short as possible, but during testing they may last for up to 400 ms (unless faults are detected).

In case of an open loop, i.e. no TE connected (or the loop termination in the TE is very highohmic), the DC voltage at the NTP will be between 36 and 66 V. Normally the voltage shall be between 42 and 66 V, but in exceptional cases, e.g. when the mains voltage has been fallen out, the voltage at the NTP may be somewhat less.

For the loop signalling, as applicable on analogue subscriber lines, the PSTN exchange supplies feeding voltage to the line and the TE provides a

passive DC loop termination; the value of the loop current is a measure for the termination resistance in the TE.
The internal DC impedance of the feeding voltage supply is dependent of the line condition. In the idle and the locked out condition, the impedance can lie between 700 Ω and 4 kΩ. In the occupied condition, i.e. during call establishment phase, communication phase and release phase, the impedance can lie between 700 Ω and 1100 Ω.; If an exchange is applied with loop current regulation, the impedance can be between 100 Ω and 2 kΩ.
When a TE seizes the line for originating a call by presenting a lowohmic loop termination, it will last some time till the exchange will have detected the seizure and switched to the feeding voltage as applicable for the occupied condition; as soon as dial tone is received, it can be assumed that the occupied condition is reached.

The contribution of the subscriber line to the total loop resistance is dependent of the used type of the cable and the length of the line; the line resistance shall normally have a value between 0 and 1000 Ω.

Because of the different parameters, which have influence on the loop condition, it is not enough to define only the values of voltage and internal resistance of the DC voltage supply (U_1 and Z_1 in figure II-4/2).
In figure II-4/3, the DC characteristic at the NTP for the line in occupied condition is defined; the working areas for the highohmic and lowohmic loop conditions of connected TEs are shown. For the line in occupied condition the DC impedance of the Infrastructure R_{IS} (feeding source plus line) is between 700 and 2100 Ω.

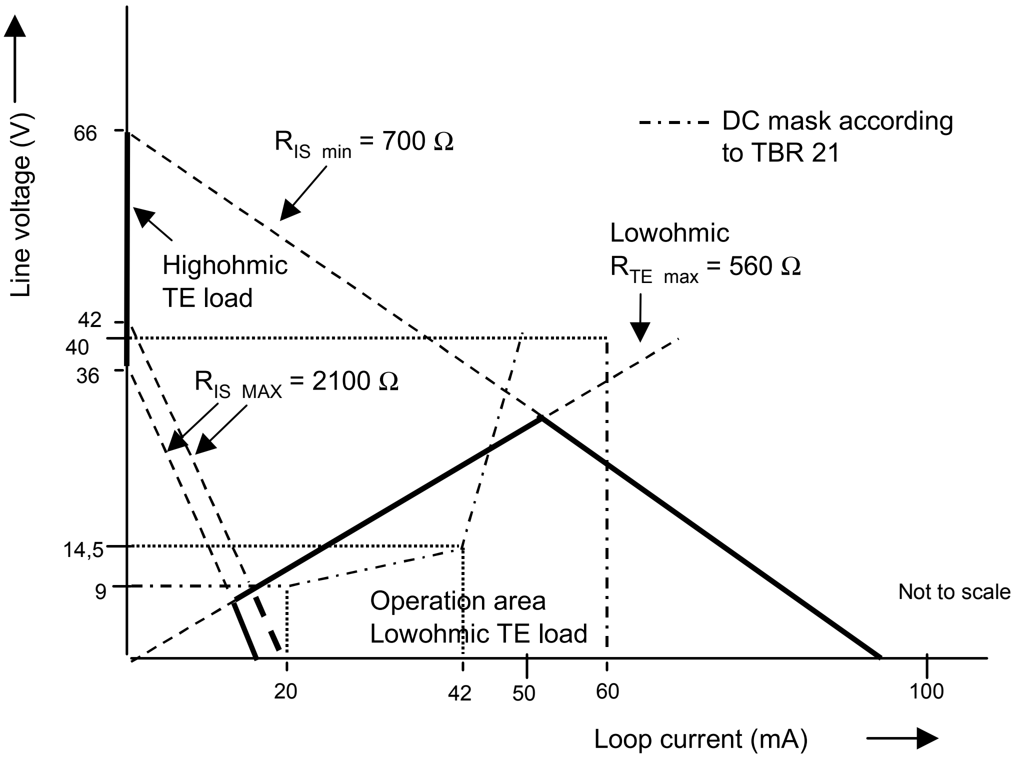


Figure II-4/3 DC characteristic at the NTP

Both the DC characteristics for TE in loop condition according to the former T 11-series requirements and according to TBR 21 are shown in the figure. The upper current limit in the TBR 21 characteristic is a specific limitation due to one European PSTN and is not applicable for the PSTN of KPN Telecom.

The application of continuous DC signals for signalling and the related timing considerations, e.g. recognition time of signals, are dealt with in the descriptions of the analogue signalling for SLTs and MLTs (chapters 6 and 7 respectively).

4.6.3 Pulsed DC signals

Pulsed DC signals are only applicable in the direction from TE to PSTN exchange; these signals are generated by changing the DC loop termination in the TE.

The following pulsed DC signals are identified:

- Pulsed dialling (Decadic pulse dialling; sometimes also called 'loop disconnect dialling' or 'loop pulsing');
- Hookflash signal;
- MCID signal.

These signals are further defined and explained in the following.

Decadic pulse dialling

Decadic pulse dialling is performed by a TE in the 'loop state', i.e. with a lowohmic loop termination, by transmission of digits in the form of series of loop interruptions (break pulses), in which the interruptions are separated by loop closures (make pulses). The loop closure or lowohmic condition between the series interruptions for a digit and the series interruptions for the next digit is defined as the interdigit pause. A break pulse and the consecutive make pulse, not being an interdigit pause, is defined as 1 dialling pulse. The number of dialling pulses of each sequentially transmitted digit corresponds to the digits 1 to 9, and 10 dialling pulses corresponds to the digit 0.

The dialling pulses shall have the following characteristics:

a)	Dialling frequency:	10 Hz ± 1 Hz
b)	Break pulse:	61,5 ms ± 10 ms; threshold: $I_{DC} \leq 8$ mA
c)	Make pulse:	38,5 ms ± 7,5 ms; threshold: $I_{DC} \geq 8$ mA
d)	Break pulse DC current:	≤ 0,5 mA during ≥ 40 ms
e)	Make pulse DC current:	≥ 14,5 mA during ≥ 25 ms

The interdigit pause is the time between the end of the last break pulse ($I_{DC} \geq 8$ mA) of a pulse train of a digit and the beginning of the first break pulse ($I_{DC} \leq 8$ mA) of the pulse train of the next digit; this pause shall be ≥ 400 ms

Loop pulsing can be accompanied with oscillations. These oscillations should be limited in order to avoid false detection of pulses. If during dialling an inductance of 4 H is included in the loop and the transient voltages due to the loop interruptions are then limited to values ≤ 140 V, then the oscillations will not disturb the dialling.

Hookflash signal

The hookflash signal is only applicable on a Single-Line Termination for which Call Waiting Hookflash (CWH) is subscribed to.
The hookflash signal is a short loop interruption (highohmic loop termination in the TE), which is used in procedures related to the CWH supplementary service. It is shorter than the call release signal. It indicates to the exchange that a waiting call (indicated with the call waiting tone) shall be through connected to the subscriber line (answering the second call); the first call is then placed 'on hold'. With subsequent sending of the hookflash signal, the user can switch between one and the other call.

The exchange will, during the active state of a call, recognize a highohmic loop termination as a hookflash signal when the highohmic loop termination is > 17 kΩ and the duration is between 90 and 800 ms.
It should be noted that from a TE with a Register Recall (RR) button, with which a loop interruption (RR-signal) between 90 and 130 ms is generated, the hookflash signal can be sent by pressing the RR button.

MCID signal

The MCID signal is only applicable on a Multi-Line Termination for which the use of MCID is activated by KPN Telecom.
Note: MCID on an SLT is not performed with a special MCID signal, but with a special dialling procedure.
The MCID signal is a short loop interruption (highohmic loop termination in the TE (PBX)), which is used to instruct the exchange that the source of an incoming call, i.e. the calling line identity, shall be identified and registered in the exchange.

The exchange will, during the active state of a call, recognize a highohmic loop termination as an MCID signal when the highohmic loop termination is > 17 kΩ and the duration is between 100 and 200 ms.

4.6.4 AC signals

AC signals with frequencies below 300 Hz, are only applicable in the direction from the PSTN exchange to the TE.

The exchange supplies the following AC signals to the NTP:
- Ringing signal;
- Metering pulses.

These signals are further defined and explained in the following.

Note:
The 50 Hz 'common mode' metering facility is no longer commercially available. The present version of the publication does contain the technical information about the facility, because the facility is still present in the network.

Ringing signal

Ringing signal is, together with a polarity reversal of the feeding voltage, supplied on network terminations as an indication to the connected TE(s) that the PSTN has seized the line while offering a call; on MLTs with the DDI feature, the ringing signal is not supplied. In case of an SLT with activated CLIP supplementary service, the supply of the ringing signal is postponed until the procedure of sending the calling line identity information with DTMF signals is completed; the seizing of the line before the sending of the DTMF signals for CLI is noticeable by the polarity reversal of the feeding voltage.

The ringing voltage signal is superimposed on the DC feeding voltage of the line. The level of the ringing voltage at the NTP is dependent on the load impedance for ringing signals of the subscriber's premises; it can be expected that, according to the European harmonized technical requirements for TEs and to the installation rules as applied in the relevant ETSI Guide, the level is at least 30 V.

The following characteristics for the ringing voltage apply:

Frequency:		23 - 27 Hz
Level:		30 - 90 V
Cadence:	on:	750 - 1250 ms
	off:	3600 - 4400 ms

For the ringing signal, a distinction has to be made between the 1st ringing voltage pulse and the periodical ringing voltage. The exchange may provide a dedicated 1st ringing pulse, a pre-ringing pulse, before periodical ringing is supplied; in this case, the ringing-off condition between the pre-ringing pulse and the 1st pulse of the periodical ringing may be short or even absent. This may result in a noticeable 1st ringing pulse with a duration of up to 2500 ms.

Metering pulses

Metering pulses are supplied to network terminations, SLTs as well as MLTs, to which the metering facility is provided by KPN Telecom. The number of pulses supplied during a basic call is an indication about the costs for that call.
A metering device or a metering function in a TE is needed for detecting metering pulses and presenting the amount of pulses for the user's benefit.
In the following, technical information on the metering system is provided.

The metering pulse is a pulsed AC voltage signal which is applied in common mode, i.e. in equal phase with respect to the PSTN network earth, to both the a- and the b-wire. For this aim the PSTN network earth is provided via a third wire to the network termination. The network provided earth connection should be used for detecting metering pulses; if another earth reference is used, there may be differences in earth levels which may cause bad functioning of the equipment.

The characteristics of the signal are:

Frequency:	48 - 52 Hz;
Pulse duration:	70 - 200 ms.

The voltage level at the NTP depends on the load impedance for 50 Hz between the a- and b-line terminals together and the network earth; for a resistive or capacitive load impedance with an absolute value between 5 kΩ and 100 kΩ, the level will be in area 'D' of figure II-4/4.
The value of the total 50 Hz common mode current, that is the current trough the a- and the b-line terminal together, shall be less than or equal to 15 mA.

A metering device should have an attenuation in the voice band (300 - 3400 Hz) of less than 0,2 dB; however in the band 300 - 500 Hz, an attenuation of up to 0,4 dB is acceptable.
The symmetry for 50 Hz common mode voltages shall be better than 46 dB.

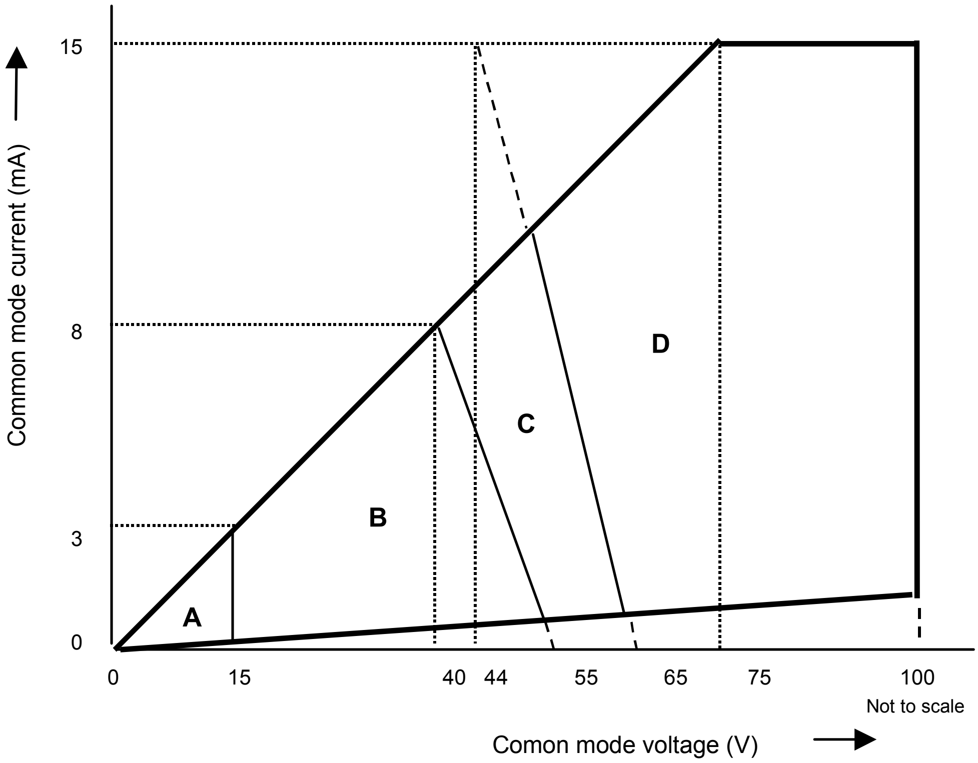


Figure II-4/4 Common mode current to voltage characteristic for metering

In figure II-4/4 the 50 Hz common mode current to voltage characteristic for resistive and capacitive loads between a- and b-line terminals together and the PSTN earth with values between 5 kΩ and 100 kΩ is provided. It is recommended that the detection level for the leading edge of the metering pulse lies in area 'C', and for the trailing edge in area 'C' or area 'B'.

The metering pulses are separated by pauses; the length of the pauses is dependent of the applicable tariff for the call.
The system is defined such that the maximum number of metering pulses per second can be three. It is recommended that a 50 Hz common mode signal with a duration shorter than 50 ms is not interpreted as a metering pulse, and that an interruption of less than 5 ms in the 50 Hz common mode signal is not interpreted as a metering pulse pause.

In the foregoing, no notice is given to possible parallel connection of metering devices to the NTP. In practice it shall normally be possible to connect two such devices in parallel to an NTP if the total impedance will remain in area 'D' of figure II-4/4.

4.6.5 Tone code signals / DTMF dialling

The tone code signals are the signals in the 3,1 kHz voice band, as defined for the DTMF signalling or dialling system, for the transport of digits or coded commands from the TE to the PSTN exchange and vice versa. The use of the DTMF signalling system in the communication phase of a call for end-to-end 'data transport' is outside the scope of this document.

The DTMF system is extensively specified in international standards, original in CEPT standard T/CS 46-02, now in ETSI Standards ES 201 235, parts 1 to 4. Part 1 of this multi-part standard specifies the general principles and coding of the DTMF system; part 2 deals with DTMF transmitters; part 3 deals with the DTMF receiver for local loop signalling and part 4 deals with receivers for use in terminal equipment for end-to-end signalling.

The application of the DTMF system in the PSTN exchanges conforms to the mentioned ETSI standard. This applies for both the DTMF receivers for receiving DTMF digits (including coded commands, e.g. the '∗' and '#', for control of supplementary services) and the DTMF transmitters for transmitting DTMF codes for DDI digits and the calling line identity for the CLIP service.

4.6.6 Audio signals / network tones

In table II-4/1 the characteristics of the network tones, as supplied by the PSTN of KPN Telecom, are provided. The frequencies, levels and cadences with applicable tolerances, are given. The levels are those which can be expected when the line at the network termination point is loaded with a resistance of 600 Ω. The actual level at the NTP is dependent of the length of the line and the load impedance at the NTP.

Table II-4/1 Characteristics of network tones

Tone	**Frequency (Hz)**	**Level (dBm)** (Note 1)	**Cadence**	
			on (ms)	**off (ms)**
Dial tone	400 - 450	-25,7 to -5,0	Continuous	
Special dial tone	400 - 450	-25,7 to -5,0	450 - 550	35 - 75
Ringing tone (Note 2)	400 - 450	-25,7 to -5,0	750 - 1250	3600 - 4400
Busy tone	400 - 450	-25,7 to -5,0	400 - 600	600 - 400
Congestion tone	400 - 450	-25,7 to -5,0	200 - 300	300 - 200
Information tone (Note 3)	900 – 1000 1350 – 1450 1750 - 1850	-31,7 to -9,8 -31,7 to -9,8 -31,7 to -9,8	260 – 400 260 – 400 260 - 400	max. 30 max. 30 max. 30 pause: 750-1250
Release tone	Identical to congestion tone			
Call waiting tone	400 - 450	-31,7 to -9,8	450 - 550	9200 - 9800
Acceptance tone	Successful supplementary service control action: Identical to dial tone; eventually, in case of CFU, special dial tone			
Rejection tone	Unsuccessful supplementary service control action: Identical to congestion tone			

Note 1:
In telephone transmission techniques it was common use to relate requirements to a reference impedance of 600 Ω resistive. To day, a complex reference impedance has been standardized by ETSI. Because of that, it is found more appropriate in ETSI documents to express values of voltage levels in the unit 'dBV', which means a level relative to a voltage level of 1 V.
For 600 Ω, the following rough approximation for the relation between the voltage level expressed in 'dBm' and expressed in 'dBV' can be applied: The value in 'dBV' is the value in 'dBm' minus 2.

Note 2:
The pause between the 1st and the 2nd ringing tone pulse may be shorter than the normal 'off' condition, or may even be absent; this may result in a 1st ringing tone pulse of 2500 ms.

Note 3:
The difference in level between any two of the three frequencies is less than 3 dB.

4.6.7 Disturbing signals

It is possible that disturbing signals are introduced on the line due to interference with external sources, e.g. strike of lightning, short-circuiting with the mains and with high tension cables, signals induced from other lines.

Strikes of lightning in the ground near the cable and short-circuiting with high voltage sources can cause very high voltages on the subscriber line. These voltages can be conducted to both the exchange and the NTP. The line circuitry in exchanges are provided with protection measures for such high disturbing voltages. At the NTP side protection measures in TE, related to the safety of users, is mandatory according to the law, i.e. the European Low Voltage Directive applies; protection measures in the circuitry of the TE is the responsibility of the TE manufacturer/supplier.

Induction of disturbing signals from other lines is mainly caused by bad symmetry to earth conditions; the asymmetry may be caused by the cable, by the PSTN exchange or by the TE or the subscriber premises wiring. With respect to this, it is important that the requirements for symmetry to earth are fulfilled by all connected equipment.

5 Terminal Equipment, PSTN network terminations

5.1 Introduction

A telecommunication infrastructure, in the case at issue the PSTN, is only of value if appropriate Terminal Equipment (TE) is available to connect to it and to support the provided services for the benefit of the user's telecommunication needs.
According to the former European legislation, applicable essential requirements for basic telecommunication services are worked out and published by ETSI as part of the telecommunication regulations, i.e. ETSI TBR 21, ETSI EN 301 437 ("TBR 37") and ETSI TBR 38. In addition ETSI has published some guidance documents, i.e. ETSI EG 201 121 on the application of TBR 21 and ETSI EG 201 120 on series and/or parallel connection aspects of TEs at PSTN Network Termination Points.

In the former technical regulations only the requirements are provided, which are identified to be essential for the basic telecommunication service, e.g. requirements related to tone detection by automatic calling and answering TEs or parallel connectivity at an NTP were not defined to be essential. Also requirements related to the support of supplementary services or facilities, which do not affect the basic service, were not essential from the regulations point of view. Such matters are still of great importance for TE, which is intended to be connected to the PSTN, and for the users of the PSTN.

In the following section 5.2, the situation with respect to the technical requirements for the PSTN TEs, as published by ETSI, is highlighted.
In section 5.3, information related to subscriber premises installation, i.e. parallel and series connection matters, is provided.
Finally in section 5.4, a number of additional access requirements for TE with automatic calling and/or answering function is provided, which may be fulfilled by TE on a voluntary basis with the aim to provide a better performance for the user.

5.2 Technical requirements

Originally the PSTNs in Europe are developed independent of each other. The technical properties at the network terminations were diverse and because of that, the applicable attachment requirements for terminal equipment were different.
In an early stage of European regulations ETSI has made an extensive inventory of the different attachment requirements in Europe and published these in ETSI EN 300 001.

Subsequently the European Commission requested ETSI to produce harmonized PSTN standards on the basis of the former TTE Directive (91/263/EEC, in 1998 succeeded by 98/13/EC).
The harmonized standards as drafted by ETSI are called TBRs (Technical Base for Regulation). Under the operative R&TTE Directive 99/5/EC, these documents have no relevance any more for technical regulation, but they are maintained to be published in one or another form by ETSI as European or ETSI standards.

At present the following PSTN TBRs are published by ETSI (only short titles are given; the full titles can be found in the reference list in section 1.3):

- ETSI TBR 21: European analogue 2-wire PSTN access standard for 'non-voice' TE;
- ETSI "TBR 37": European analogue 2-wire PSTN access standard for 'voice' TE (published by ETSI as ETSI EN 301 437);
- ETSI TBR 38: European PSTN standard for TE with analogue handset functionality for the voice telephony justified case service.

TBR 21 and "TBR 37" contain the access requirements for interworking with the PSTN; TBR 38 contains the voice telephony requirements for interworking via the network for end-to-end quality of the telephony service.

For the distinction whether a TE is of the category 'non-voice' TE or 'voice' TE, the scope statements of TBR 21 and TBR 37, as imposed by the European Commission, should give guidance.
The scope statement of TBR 21 declares that the TBR is applicable for all TE which is capable of 2-wire analogue PSTN access, "excluding TE which is capable of supporting the voice telephony justified case service as specified in the former TTE Directive".
The scope statement of TBR 37 declares that the TBR is applicable for TE with the 2-wire analogue PSTN access capability, which is intended "for supporting any voice telephony service".

The Commission's request for TBR 21 was the first step to harmonized PSTN access requirements in Europe.
Because of technical differences in the PSTNs in Europe, and also because of different interpretations of the judgement criteria for the essentiality question, a lot of discussions were needed, both in the technical group of ETSI and in European regulatory committees. In order to progress, the Analogue Type Approval Advisory Board (ATAAB) was set up with the aim to draft Advisory Notes (ANs). An AN may deal with a requirement, which is general applicable for all networks, or may deal with a specific requirement for a distinct network or country, which is then mentioned by name. For the time being the ATAAB Advisory Notes are published in annexes to ETSI Guide EG 201 121: "A guide to the application of TBR 21". Also these ANs have lost their relevance for regulation, but remain to be of value as additional technical information. Discussions are going on within ETSI for safeguarding this information for the future, e.g. by adapting the ANs into ETSI Technical Specifications or Technical Reports. Most of the related information can also be found in ETSI EN 300 001.

With respect to the foregoing, it can be seen that KPN Telecom has not imposed any specific Advisory Note for The Netherlands. KPN Telecom is of the opinion that aspects of interworking with the network, which are not covered in TBR 21 (or "TBR 37"), and for the support of services and/or facilities other than voice telephony, are for the responsibility of the TE manufacturer and/or supplier on a voluntary basis.
The intention of KPN Telecom with the present publication is to present relevant information about the PSTN network terminations of KPN Telecom for the benefit of the TE manufacturers and suppliers.

With respect to the European harmonized PSTN access standards TBR 21 and "TBR 37", specific exclusions are created for:
- dialling methods;
- rules for parallel and series connection.

Dialling methods
According to the scope statements of the TBRs, these apply for TE which, when capable for originating calls, use DTMF signalling; other signalling methods, e.g. decadic dialling (loop disconnect dialling), if provided in the TE and intended to be used for a specific PSTN in a country, shall be in accordance with the related specification. The specification for decadic dialling for interworking with the PSTN of KPN Telecom can be found in section 4.6.3 of the present publication.

The PSTN of KPN Telecom supports DTMF signalling for all network terminations; so, for TE intended to be connected to the PSTN of KPN Telecom, there is no need to support decadic dialling. KPN Telecom is in favour of the use of DTMF dialling as it gives a much better performance for the user.

Rules for parallel and series connection
According to the scope statements of TBR 21 and "TBR 37", the requirements of the TBRs apply for a single TE connected to the PSTN network termination.
The meaning of this is that a TE is allowed to consume all of the available network capabilities at the access point; customer's TE configuration matters, i.e. parallel and/or series connection of TEs, were excluded from the regulations.
The European Commission did recognize the importance of connectivity rules for the PSTN customers and did request ETSI to draft a guide on installation matters for PSTN network terminations. As the result of this request, ETSI has published ETSI Guide EG 201 120: "PSTN; Method of rating TE so that it can be connected in series and/or in parallel to a Network Termination Point".
Detailed information on these installation matters is provided in section 5.3.

Further, a number of access requirements can be identified, which may be fulfilled by TE on a voluntary basis. For example, TE with automatic calling and answering function may make use of the properties of the PSTN of KPN Telecom; in section 5.4 some information is provided.

General information about supplementary services or facilities is provided in chapter 2. Specific access requirements with respect to the supplementary services "Calling Line Identification Presentation" (CLIP; Dutch: 'Nummerweergave') and "Call Waiting Hookflash" (CWH; Dutch: 'Wisselgesprek') are presented in related sections in chapter 6 as part of the description of the signalling for PSTN Single-Line Terminations (SLTs).

5.3 Installation matters; ETSI Guide 201 120

Note: When in this section TBR 21 is mentioned, it is intended also to refer to "TBR 37".

5.3.1 Introduction

As mentioned in the foregoing, the requirement values in TBR 21 are related to the application of a single TE to be connected to the PSTN 2-wire analogue network termination; a TE is allowed to consume all the available network capabilities at the NTP.
Furthermore, the subject of series connection of TE at the customer's premises is not covered in the TBRs.

In practice, customers will have the need to connect more than one TE to there PSTN Network Termination Point (NTP); this may be an arbitrary combination of several TEs connected in parallel and/or in series.
In order to give guidance on this matter to all involved parties, i.e. the network operator, the terminal manufacturer/supplier and the PSTN customer, the European Commission did request ETSI to draft and publish a document on these installation matters. This has resulted in the publication of ETSI Guide EG 201 120: "PSTN; Method of rating TE so that it can be connected in series and/or in parallel to an NTP".

First point to note is that in case of a single connected TE, there can not be any negative influence from other TE on the communication performance, because no other TE is connected to the line. But when other TEs are parallel and/or series connected, the performance of a TE in communication state may be degraded by the qualities of the other TEs in quiescent state and/or in loop transferred state; especially for the voice telephony service, this is not acceptable.

In EG 201 120 the requirements and limits are identified, which are important in relation to parallel and series connection. Some of these requirements are not contained in TBR 21; in that case limits and references to test methods, based on the information contained in EN 300 001, are provided in the guide.
Detailed information on the parallel aspects of parallel and series connection of TEs is provided in following subsection 5.3.2.

For characterizing parallel and series connection, EG 201 120 defines for relevant parameters an arbitrary loading scale and provides a formula for calculating the actual related "loading factor" of a TE.
This approach is in principle the same as the former "connection factor" (Dutch: "Aansluitfactor") system as defined in the 'old' national technical regulations T 11-00 and T 11-01; the system of the "connection factor", as was applicable in The Netherlands, is also defined in EN 300 001, clause 2.2.1.
Important difference between the NL "connection factor" and the ETSI "loading factor" system is, that the ETSI system is based on 9 parameters while the Dutch system is based on only 3 parameters. Besides that, the used arbitrary scales are different. Detailed information on this subject is provided in subsection 5.3.3.

Subsection 5.3.4 deals with additional parameters for series connected TE in 'loop transferred state' or in 'through connected quiescent and loop state'.

5.3.2 Parallel aspects of parallel and series connection

In table 1 of ETSI Guide EG 201 120 the parallel aspects of parallel and series connection are highlighted and the relevant parameters are given. In addition to the requirements according to TBR 21, some other parameters are essential for TE, which is intended to be connected in parallel and/or in series with other TE to a PSTN NTP. For these additional requirements, reference is made to EN 300 001, in which similar requirements are contained.

For series connected TE, the requirements apply in quiescent state with the series port open (or with a load resistance of 1 MΩ, representing a secondary connected TE in quiescent state, connected to the series port). Some requirements apply also for series connected TE in 'loop transferred' or 'through connected loop' state for 'loop transferring TE' and 'through connecting TE' respectively; see subsection 5.3.4 for the definitions of these types of series connected TE.
Where in the following 'loop transferred state' is mentioned, also the 'through connected loop state' for through connecting TE is meant.

In Table II-5/1, which is a slightly modified copy of Table 1 of the ETSI guide, an overview of and rationale for the relevant parameters is presented.
The parameters concerned are:

1) Resistance to earth in quiescent and loop transferred state (TBR 21: clause 4.4.4):
   The total presented DC resistance between each wire of the 2-wire line interface of the TE at the NTP and earth shall be sufficiently high.
2) Impedance to earth at 50 Hz in quiescent and loop transferred state (EN 300 001: clause 9.2.2.1):
   This parameter was only of relevance for the 50 Hz common mode metering facility; this facility is no longer commercially available.
3) DC resistance in quiescent state (TBR 21: clause 4.4.1):
   The total presented DC resistance between the wires of the 2-wire line interface of the TE at the NTP shall be sufficiently high.
4) Impedance at 25 Hz and 50 Hz in quiescent state (TBR 21: clause 4.4.2.1):
   The total presented impedance for ringing signals at the NTP shall be sufficiently high. Beside the requirement for the impedance at the ringing frequency, TBR 21 contains a requirement for the transient response to ringing signals, see TBR 21, clause 4.4.2.2. The rationale for this requirement is that the TE should represent a sufficient low RC time for transients during ringing in order not to create a false ring trip detection (unintended detection of the answered condition) in the PSTN exchange. It is recognised that, if each individual TE fulfils the transient response requirement, the requirement is fulfilled for the installation with a number of parallel connected TEs too.
   Note that the PSTN of KPN Telecom does only provide 25 Hz ringing.
5) Voice band impedance in quiescent state (EN 300 001: clause 4.1.1):
   If a TE is in communication condition and the total presented voice band impedance of parallel connected TE in quiescent state is to low,

then this parallel TE will create a substantial signal loss (parallel connection loss).

Table II-5/1 Table 1 from EG 201 120, modified

"Table 1: Parallel aspects of parallel/series connection"

Operating state of TE	**Parameter**	**Test method**	**Value for 100 LU**	**Formula for calculation of LF and unit for input data;** See note 4
Quiescent, Transferred	Lowest resistance to earth	TBR 21: A.4.4.4	R = 10 MΩ	1 000/R [MΩ]
Quiescent, Transferred	Lowest impedance to earth at 50 Hz See note 1	EN 300 001: A.9.2.2.1 Test values: V_f = 50 V, R_f = 1200 Ω U=100V_{rms}, Z_G =1400Ω Z_L = 600 Ω	Z =200 kΩ	20 000/Z [kΩ]
Quiescent	Lowest DC resistance	TBR 21: A.4.4.1	R = 1 MΩ	100/R [MΩ]
Quiescent	Lowest impedance at 25 Hz and 50 Hz	TBR 21: A.4.4.2.1	Z = 4 kΩ	400/Z [kΩ]
Quiescent	Lowest impedance in the range 0,3 - 3,4 kHz	EN 300 001: A.4.1.1 Test values: V_f = 50 V, R_f = 1200 Ω V_{t2} = 1,0 V_{rms}	Z = 10 kΩ	1 000/Z [kΩ]
Quiescent	Lowest impedance at 12 kHz and 16 kHz ± 1 % See note 2	EN 300 001: A.4.1.1 Test values: V_f = 50 V, R_f = 1200 Ω V_{t2} = 1,0 V_{rms}	Z = 10 kΩ	1 000/Z [kΩ]
Quiescent	Maximum DC current during ringing	TBR 21: A.4.4.2.3	I = 0,6 mA	$100 * I / 0,6$ [mA]
Quiescent, Loop	Lowest unbalance loss about earth in the range 50 – 3400 Hz	TBR 21: A.4.4.3 and A.4.7.4	LCL= 46 dB	$100 * 10^{(46-LCL)/20}$ [dB]
Transferred	Lowest unbalance loss about earth in the range 50 – 3400 Hz	EN 300 001: A.4.2.2.2 Test values: V_f = 50 V, R_f = 1200 Ω R_L = 360 Ω, C_L = 20 µF, L = 10 H See note 3	LCL= 46 dB	$100 * 10^{(46-LCL)/20}$ [dB]
Quiescent, Transferred	Maximum noise	EN 300 001: A.4.5.1 Test values: V_f = 50 V, R_f = 1200 Ω Z_L = 600 Ω	N=-64 dBmp	$100 * 10^{(64+N)/10}$ [dBmp]

Note 1: Only applicable for terminals providing a common reference terminal which is connected to the PSTN earth and which is not intended to detect metering pulses.
Note 2: This applies for a TE not intended to detect metering pulses. A TE intended for detection of metering pulses has typically an impedance of nominally 200 Ω within the detection frequency band, see EN 300 001, clause 9.2.1.4.
Note 3: Measured with 600 Ω load simulating the AC load of a terminating TE in loop state.
Note 4: R, Z, I, LCL and N are the measured values of the TE.

6) Impedance in quiescent state at 12 and 16 kHz (EN 300 001: clause 4.1.1):
The rationale for this parameter is related to the application of 12 or 16 kHz metering pulse systems. Note that this parameter is not relevant for connection to the PSTN of KPN Telecom as this network does not provide such type of metering signals.

7) DC current during ringing in quiescent state (TBR 21: clause 4.4.2.3): Asymmetric load of the ringing signal by TE may cause a resulting DC current. The total resulting DC current at the NTP during ringing shall be sufficiently low in order to prevent for false ring trip detection (unintended detection of the answered condition) in the PSTN exchange.
8) Unbalance about earth in quiescent, loop and loop transferred state (TBR 21: clause 4.4.3, TBR 21: clause 4.7.4 and EN 300 001: clause 4.2.2.2 respectively):
   On analogue 2-wire PSTN lines, the symmetry of the wires with respect to earth shall always be maintained in order to avoid crosstalk.
9) Noise in quiescent and loop transferred state (EN 300 001: clause 4.5.1):
   Parallel and/or series connected TE shall not introduce inconvenient signals during communication.

For rating the connection capability of TEs in a parallel and/or series installation at an NTP, the following two new terms are introduced in the ETSI Guide:

Loading Factor (LF): The portion of PSTN resources used by a TE or a set of TE (installation) when connected to a Network Termination Point (NTP).

Loading Unit (LU): An arbitrary unit to measure (or evaluate) the Loading Factor.

In TBR 21, the European limit values for parameters are defined, e.g. the DC resistance shall be not less than 1 MΩ, the DC current during ringing shall be less than 0,6 mA. The limit values represent the minimum or, if applicable, the maximum value which is allowed to be connected at the NTP; a TE is allowed to consume all of the available capacity. The European limit values are the harmonised values which must at least be supported by a European PSTN; the limit values of real PSTNs, e.g. the PSTN of KPN Telecom, may differ from the European values.

For the parameters, which are not contained in TBR 21, reference is made to EN 300 001; the applicable European limit values are defined in Table 1 of the ETSI Guide; see Table II-5/1.

It is arbitrary defined that the above mentioned European limit values are representing 100 Loading Units (LU). A TE, which fulfils exact the European limit value of a parameter, has an Loading Factor (LF) of 100 LU for that parameter.
If a TE has a value for a parameter which is higher or, if applicable, lower than the limit value, then the LF is less than 100 LU.

Note:
This arbitrary defined unit is further indicated as LU_{ETSI}.; this means that in Table II-5/1, the unit LU should be read as LU_{ETSI}. In subsection 5.3.3, the specific Dutch unit LU_{NL},, which is based on the arbitrary scaling according to the 'old' Dutch 'connection factor' (see EN 300 001), is defined.

The following example explains the methodology of the loading factor system.

If the DC resistance of a TE is 2,3 MΩ, then the LF for the DC resistance of that TE is:
1 MΩ / 2,3 MΩ x 100 LU_{ETSI} = 43,4... LU_{ETSI},, rounded up to 44 LU_{ETSI}..

If this TE is connected to an NTP, there can another TE be parallel connected which has an LF for the DC resistance of up to 100 - 44 = 56 LU_{ETSI}; this is a TE with a minimum DC resistance of:
100 LU_{ETSI} / 56 LU_{ETSI} x 1 MΩ = ± 1,8 MΩ.

The chosen arbitrary scale of 100 gives the impression that the system is exact.
In practice, the parameter values of a type of TE shall have tolerances because of the tolerances in the used components for that type of TE. So also the LF of that type of TE shall have a comparable tolerance; e.g. if the mean DC resistance of a type of TE is 2,3 MΩ and the tolerance for that product is 5 %, then the LF for the DC resistance of the TE can have a value between 42 and 46 LU_{ETSI}.
Furthermore, the available network capability at an NTP is dependent of tolerances in the PSTN exchange and of the quality and length of the subscriber line.

So the system can not be very exact. But the system gives useful practical indications for suppliers and there customers in order to evaluate the expected performance of the customer's installation related to the available network capability.

In the ETSI guide, the supplier of a TE is asked to make the LF of the TE (expressed in the unit LU_{ETSI}) readily available to the user. The published LF value should be the highest of the different LF values as calculated for the individual parameters in Table II-5/1, rounded up to the nearest whole number. In addition, the supplier may also provide the applicable LF values of all the individual parameters.

The network operator is asked to publish the LF that the PSTN can support at an NTP. This LF value should be the lowest of the different LF values for the individual parameters of Table II-5/1, rounded down to the nearest whole number. In addition, the operator may also provide the applicable LF values of all the individual parameters.
Information about the network capability of the PSTN of KPN Telecom is provided in following subsection 5.3.3.

5.3.3 ETSI "Loading Factor" and NL "Connection Factor"

As stated before, the ETSI "Loading Factor (LF)" system and the 'old' "Connection Factor (CF)" (Dutch: "Aansluitfactor") system in The Netherlands are in principle the same, i.e. for a TE is determined how much of the available network capability is consumed by the TE.
In both systems an arbitrary scale is defined; in the ETSI system the European available network capability is defined to be 100 units, while in the NL system the NL available network capability is defined to be 5 units. Because of this difference in scaling, the units are hereafter indicated as ETSI Loading Units (LU_{ETSI}) and NL Loading Units (LU_{NL}) respectively.
So, in the ETSI Loading Factor system the LF values are expressed in LU_{ETSI} and in the NL Connection Factor system the CF values are expressed in LU_{NL}.

In the NL CF system as defined in EN 300 001, clause 2.2.1, the available network capability at the NTP of the PSTN of KPN Telecom is defined to be 5 LU_{NL}. The CF of a TE can have a value in the range 0,5 LU_{NL} - 1,0 LU_{NL} - 1,5 LU_{NL} - etc. The maximum value is 2,5 LU_{NL} which allows at least 2 TEs to be connected in parallel at an NTP.
The NL CF system is not applicable for PBXs; a PBX is allowed to use the total available network capability (= 5 LU_{NL}).

In the ETSI LF system, the available network capability, which European PSTNs should at least support at the NTP, is defined to be 100 LU_{ETSI}.
The LF of a TE can have a value which is the highest of the calculated values according to Table I-5/1, rounded up to the nearest whole number.

The question is, what is the available network capability at the NTP of a real PSTN, e.g. the PSTN of KPN Telecom, expressed in the number of LU_{ETSI}; this number should be at least 100.

In the ETSI LF system, the LF is defined for 9 parameters according to Table II-5/1 (the parameter 'lowest unbalance about earth' is contained in the table twice because of the different test methods).
In the NL CF system only 3 of these 9 parameters, i.e. 'DC resistance', 'lowest impedance at 25 Hz' and 'lowest impedance in the voice band range', are defined to be decisive for evaluating the loading at an NTP and for determining the CF.
Note that related to the 25 Hz ringing signal, a combination of the 'impedance at 25 Hz' and the 'capacity of the TE in quiescent state' is used in the NL system; in stead of the requirement for the capacity of the TE in quiescent state, TBR 21 contains a requirement for the transient response to a DC excitation of the TE.
For the other parameters, the requirement values in the 'old' Dutch regulations as contained in EN 300 001 were such that, for an acceptable quality of the performance, at least 5 TEs can be connected in parallel at the NTP.

Evaluation of the available network capability at an NTP of the PSTN of KPN Telecom, expressed in ETSI Loading Factor terms:

1) Lowest DC resistance in quiescent state:
   The minimum required DC resistance is about 400 kΩ; this corresponds to a maximum capability for the DC resistance of 250 LU_{ETSI}.
2) Lowest impedance at 25 Hz in quiescent state:
   The minimum required impedance for the ringing signal is about 2 kΩ; this corresponds to a maximum capability for the ringing impedance of 200 LU_{ETSI}.
3) Lowest impedance in the range 0,3 - 3,4 kHz in quiescent state:
   The minimum required impedance in the voice band is about 7,5 kΩ; this corresponds to a maximum capability for the voice band impedance of about 130 LU_{ETSI}.
4) The remaining 5 parameters:
   For these parameters, no direct relation with the NL CF system can be made. The most decisive and critical parameter of these is the 'DC current during ringing in quiescent state'; for this parameter the maximum required value is about 0,75 mA, which corresponds to an LF

capability of 125 LU_{ETSI}. For the other parameters, the LF capabilities are at least 125 LU_{ETSI}.

From the foregoing it can be concluded that the general Loading Factor LF, representing the overall available network capability at the NTP of the PSTN of KPN Telecom, is 125 LU_{ETSI}.

According to the 'old' NL CF system, the general Connection Factor CF for the available network capability of the PSTN of KPN Telecom is 5 LU_{NL}.

As the result of the foregoing, the following relation can be applied for the evaluation of the subscriber's premises loading at the NTP:

CF (LU_{NL})	**LF (LU_{ETSI})**
0,5	12,5
1	25
1,5	37,5
2	50
2,5	62,5
5	125

In the NL CF system only the general CF value of a TE is published; the individual CF values for each parameter may not be available; it may also not be known which parameter was decisive for the general CF value.
This means that, when a combination of 'old' and 'new' types of TE are applied in the customer's installation, only the general LF values of TEs can be taken into account for evaluating the loading at an NTP.
In case of an installation with only 'new' types of TE, of which the individual LF values (expressed in LU_{ETSI}) are available, these individual LF values can be evaluated; the limits for the available network capabilities, as given in the points 1) to 4) above, apply.

5.3.4 Additional parameters for series connected TE

For series connected TE, some specific requirements are to be taken into account in order to guarantee an acceptable performance of the TE, which is connected at the secondary port (series port).

The following types of series connected TE can be identified:
a) **Loop transferring TE**: this type of TE is intended for transferring its loop state to a TE, which is connected at the series port;
b) **Through connecting TE**: this type of TE is intended to be connected between an NTP and a TE, irrespective whether the TE, which is connected at the series port, is in the quiescent state or in the loop state.

In Table 2 of ETSI Guide EG 201 120 the additional requirements, which are essential for series connected TE, are given; for each parameter, reference is made to the similar requirement in EN 300 001.
The requirement for the 'insertion loss at 25 Hz and 50 Hz' is additional to the requirements for insertion loss in clause 4.3 of EN 300 001; this requirement is specific for 'through connecting TE' with a connected secondary TE in quiescent state, and is related to the loading characteristic for the ringing signal.

The following general rationale for the requirements for series connected TE in 'loop transferred' or 'through connected loop' state is given in EG 201 120:
"A TE intended for series connection with other TEs will need to consume very little of the available resources when in loop transferred (or in through connected loop) state, because most of the resources are needed for the TE, which is terminating the connection and is in loop state".

The same rationale applies for the requirement related to the loading of the ringing signal by a 'through connecting TE' in behalf of a terminating TE in quiescent state.

In Table II-5/2, which is a slightly modified copy of Table 2 of the ETSI guide, an overview of the identified additional requirements is presented.

Table II-5/2 Table 2 from EG 201 120, modified

"Table 2: Recommended maximum values for series connected TEs"

Parameter	Recommended maximum value	Test method
DC series resistance	50 Ω See notes 1,2	EN 300 001: A.2.5 Test values: V_f = 50 V, R_f = 1200 Ω R_L = 360 Ω
Insertion loss at 25 Hz and 50 Hz See note 3	0,4 dB	EN 300 001: A.4.3 Test values (see also note 4): Z = Z_L = 4 kΩ at 25 and 50 Hz respectively R_L = 1 MΩ
Insertion loss in the range 0,3 - 3,4 kHz	0,4 dB	EN 300 001: A.4.3 Test values (see also note 4): Z = Z_L = Z_R R_L = 360 Ω
Insertion loss at 12 kHz and 16 kHz ± 1 %	0,4 dB	EN 300 001: A.4.3 Test values (see also note 4): Z = Z_L = 200 Ω R_L = 360 Ω
Note 1: The voltage drop should not be greater than that which would be dropped across a 50 Ω resistor replacing the TE. Note 2: Some series connected TEs may present a non-linear voltage-current characteristic, e.g. a diode bridge. The voltage drop of such devices should not exceed 2 V for loop currents not exceeding 40 mA. It is recognised that this may cause incorrect operation in terminating equipment when connected to long lines. Note 3: Only applicable for through connecting TE in through connected quiescent state. Note 4: V_f = 50 V, R_f = 1200 Ω, e = 1,0 V_{rms}, C_L = 20 µF, L = 10 H. Z_R is defined in TBR 21.		

The requirements concerned are:
1) DC series resistance in loop transferred or through connected loop state (EN 300 001: clause 2.5);
2) Insertion loss at 25 Hz and 50 Hz in through connected quiescent state;
3) Insertion loss in the voice band in loop transferred or through connected loop state (EN 300 001: clause 4.3);
4) Insertion loss at 12 and 16 kHz in loop transferred or through connected loop state (EN 300 001: 4.3). This parameter is related to the application of 12 or 16 kHz metering pulse systems and is not

relevant for connection to the PSTN of KPN Telecom as this network does not apply such type of metering system.

In the table a maximum value for each parameter is recommended under the presupposition that a concatenation of maximum two series connected TEs, terminated with a normal TE, may be present in one particular branch of an installation.

5.4 TE with automatic calling and/or answering function

For TE with automatic calling and/or answering function some additional voluntary requirements are presented hereafter; these type of TEs are in this section further indicated as 'auto-call TE' and 'auto-answer TE' respectively.

Auto-call TE: Switching after dialling
After having send out the last digit, the TE should terminate the dialling state within 1 s and go to the state in which it is capable of exchanging voice frequency signals via the PSTN.

Auto-answer TE: Ringing signal detector insensitivity
According to TBR 21, it is mandatory for a ringing signal detector of a TE to meet the sensitivity requirements as defined in the related clause.
In addition it is advised that a ringing signal detector of a TE, which has the function of automatic establishing the loop state, shall be insensitive for:
a) 25Hz AC voltages, level < 15 V_{rms}, at any cadence;
b) 25Hz AC voltage, level up to 90 V_{rms}, cadence 0,1 s on / 1 s off;
c) Polarity reversals;
d) Decadic dialling from a parallel connected TE.

Auto-answer TE: Automatic establishment of loop state
The time between receiving the ringing signal and the automatic establishment of the loop state (answered state) should be greater than 1 s in order to allow the network to send the ringing tone to the calling line for the benefit of the calling user or the calling TE.

Auto-answer TE: Answering signal
After the automatic establishment of the loop state, the TE should send an answer indication to the calling side by means of a recorded message or a tone.
If a tone is applied, this should have such a characteristic that it can not be interpreted as a network tone (e.g. dial tone, ringing tone, busy tone) in order not to confuse the calling user.
The requirements for the sending power limitations according to clause 4.7.3 of TBR 21 apply.
In addition the following requirements may be applied.

If the answer indication is a recorded message:
a) the transmission should not start within 500 ms after the establishment of the loop state;
b) the send level should be not less than -15 dBm.

If the answer indication is a tone:
a) the frequency of the tone should be in the range 800 Hz to 2200 Hz;

b) the transmission should start within 2,5 s after the establishment of the loop state;
c) the duration should be ≥ 2,5 s;
d) the send level should be not less than -15 dBm.

Auto-call TE / auto-answer TE: Automatic control of loop state
TE in loop state should automatically revert to the quiescent state when it can be assumed that maintaining the loop state makes no sense any more.
Different methods for automatic control of the loop state are identified. Each can be used on its own or in combination with one and the other method(s).

a) **TE without information-related control of loop state:**
This type of TE should revert to the quiescent state no longer than 120 s after the following event:
- For auto-call TE: after sending of the last digit;
- For auto-answer TE: after the automatic establishment of the loop state.

b) **TE with carrier or data transfer related control of loop state:**
This type of TE, e.g. modems and fax machines, should revert to the quiescent state when the ITU-T defined carrier is not received or when no data transfer took place during 60 s.

c) **TE with network tone related control of loop state:**
In case of unsuccessful calls the network does provide acoustic indications, most of them in the form of tones.
In the release phase, after a communication phase or after the phase of listening to network provided acoustic indications, the network does provide the release tone.
Information about the network provided acoustic indications can be found in the chapters 3 and 4.
TE with network tone related control of loop state may use the different network tones as indication of an unsuccessful call.
TE should ultimately revert to the quiescent state within 20 s after the start of the release tone provided by the network.
The release tone has the same characteristics as the congestion tone; the characteristics are provided in section 4.6.6.

d) **TE with DC polarity related control of loop state:**
In the signalling procedures, as applicable for the PSTN of KPN Telecom, the polarity of the line feeding voltage is reversed from Idle polarity I to Reversed polarity R in the following way (see the chapters 6 and 7 for details):
- For an originating call: As number received indication after the sending of the last digit;
- For a terminating call: As line seizure signal, followed by either the ringing signal or by DTMF signals for applying the calling party number and then the ringing signal;
- In case of line test: For measurements with reversed polarity, without applying ringing signal (only in case of manual tests).

The idle polarity I is dependent on the access network and customer's premises wiring. The idle polarity at a single line termination is not

guaranteed and may be 'a-wire positive with respect to b-wire' or 'a-wire negative with respect to b-wire'.
Because of the different meanings of the polarity reversal from I to R polarity, it is not advised to use this by auto-answer TE as an indication for an incoming call; the ringing signal is the only reliable indication.
When a call has to be released by the PSTN while the polarity on the line is still Reversed (R), either because the other party has cleared the call or some time out in the PSTN has been expired, the polarity is reverted back to the I polarity and the release tone (same characteristic as the congestion tone) is provided.
When for a line with R polarity, the polarity is reverted back to the I polarity and this polarity is remained for at least 200 ms, the TE may interpret this as network release indication and revert then, e.g. within 2 s, to the quiescent state.

6 Analogue signalling, PSTN single-line termination

6.1 Introduction

This part II about the "Analogue PSTN network terminations" contains in chapter 3 a functional description of the basic PSTN call handling process, as this is perceivable on the subscriber line of the customer.
In sections 4.5 and 4.6, general aspects of subscriber line signalling, i.e. the physical and electrical appearance of the signals, are dealt with.

In the present chapter, the detailed dynamic signalling procedures on the analogue subscriber line of a Single-Line Termination (SLT) are described, including the detailed signalling process for those supplementary services which require support by Terminal Equipment (TE), i.e. Calling Line Identification Presentation (CLIP; Dutch: 'Nummerweergave') and Call Waiting Hookflash (CWH; Dutch: 'Wisselgesprek®').
The signalling is applicable on all SLTs. TE, intended to be connected to an SLT of the PSTN of KPN Telecom and complying with the former technical regulations, e.g. TBR 21, EN 301 437, TBR 38, and with the additional requirements in the present chapter for the supplementary services in question, are expected to interwork correctly with the signalling.

The analogue SLT signalling is a DC loop signalling system for 2-wire analogue subscriber lines; the DC feeding voltage of the line is supplied by the PSTN exchange and the loop termination is applied by the TE. In general, use is made of continuous signals (line signalling states) corresponding with the actual phase of the call handling; for general information about other type of signals (e.g. pulsed signals, audio signals), see sections 4.5 and 4.6.

The continuous DC line signals from the exchange are provided in the form of the polarity of the DC feeding voltage; the polarity is further indicated as "Idle polarity (I)" and "Reversed polarity (R)".
It is not possible to guarantee for a connected TE at an SLT, whether the I-polarity is 'a-wire positive with respect to b-wire' or 'a-wire negative with respect to b-wire'. Because of installation and manipulation activities in the infrastructure or in the subscriber's premises installation, the a- and b-wire may be interchanged somewhere. This should not be a problem for TE which is intended to be connected to an SLT.
During switching actions in the exchange, short lasting feeding interruptions of up to 80 ms may occur, which have no signalling meaning; due to tests, the interruption time may be enlarged slightly.
The voltage level of a- and b-wire with respect to earth is dependent of the operative line circuit in the exchange; this level has no signalling meaning.
During test and maintenance activities related to the line, special conditions, e.g. feeding interruptions, may be experienced which have no signalling meaning; information about these situations is provided in chapter 8.

The continuous line signals from the TE (or in fact from the subscriber's TE installation as presented at the NTP) are the 'highohmic' loop termination (≥ 400 kΩ) and the 'lowohmic' loop termination (≤ 560 Ω). If in the highohmic condition in band information has to be transferred, i.e.

DTMF signals for CLIP (see subsection 6.2.3), the loop termination shall be 'medium highohmic' (between 90 and 110 kΩ).
In the active state of the call, a loop termination of > 17 kΩ will be seen as 'highohmic', i.e. for detecting the hookflash (HF) signal.
The loop state detection function in the exchange is implemented such, that the NTP presented DC loop termination of the TE installation and the influence of the subscriber line cable are taken into account.

In following sections, the signalling procedures are described in relation with the different call types, i.e. an originating call (call type O) and a terminating call (call type T).
This chapter contains in section 6.2 a general introduction, the description of some special signalling conditions and the specific conditions and procedures related to the CLIP and CWH supplementary services. The procedures related to the Completion of Calls to Busy Subscriber (CCBS) supplementary service are described in section 6.3 as part of the originating call process.
An explanation of the indications and abbreviations as used in this chapter can be found in subsection 6.2.5.

6.2 Introduction to the signalling procedures and special conditions

6.2.1 General

The detailed description of the signalling procedure for a call from TE to the PSTN, the Originating (O) call type, is provided in section 6.3, and for a call from PSTN to TE, the Terminating (T) call type, is provided in section 6.4.
The procedure for the call type O and the call type T can respectively be compared with the originating and terminating call process in the PSTN call handling procedures; see chapter 3.
At the end of each section, the state transition diagram and the tabled signalling diagram for the related call type are provided.

In the signalling tables, the applicable parameter values of the different signalling states are indicated in a symbolic way; the direction of related signals are indicated with an arrow. Also, information about the sending and recognition times and other conditions is provided.
By sending time of a continuous or pulsed signal is to be understood, the duration that the signal or the state has to be sent uninterrupted so that reliable recognition of the signal or state at the receiving side is possible. Because the duration of signals may be increased or decreased while transmitting them over a long line, the sending time values differ from the recognition time values; the deviation between sending and recognition time values is necessary for a reliable interworking between PSTN and TE.
By recognition time of a state transition is to be understood, the duration that the new state has to be present before the new state is entered and actions related to the new state are started; herewith, it is considered that the new state is present from the time that the physical condition of the line conforms to the requirements of the new state.
Subsection 6.2.5 hereafter contains, beside an explanation of used indications and abbreviations, some further explanation about sending and recognition times.

Although the analogue signalling for SLT is provided with line condition signals from the exchange to the TE by way of polarity reversals of the line feeding voltage, it is not necessary for TE to make use of these line signals. In fact, TE for PSTN access via an SLT is required to be polarity independent; there is not much need for implementing polarity detectors in such TE. For that reason, the PSTN line signals are always accompanied by another indication, i.e. SEIZ-IS by Ringing Signal (RS) and R-IS before R-TE by release tone (RLT; same characteristic as congestion tone).
For TE supporting the CLIP service, it is required to use the polarity reversal of SEIZ-IS in order to provide in time the NIT state for receiving the CLI number information; see subsection 6.2.3.
Further, the polarity reversal of R-IS may be used by automatic TE, if feasible.

6.2.2 Special signalling conditions

The following special signalling conditions are identified:

- Double seizure (call collision);
- Blocking;
- Locking out.

Double seizure

In case of both-way lines (the general situation for SLTs), it is possible that about the same time the TE sends the SEIZ-TE signal for originating a call and the exchange sends the SEIZ-IS signal for offering a terminating call; the SEIZ-TE signal and the SEIZ-IS signal "collide with each other".
In case of such double seizure, the terminating call from the PSTN has precedence; there is already a complete communication path through connected in the PSTN for such call.
The PSTN exchange has no means to detect double seizure. The implementation of the call handling in the PSTN exchange is such that a double seizure is only possible, when the time difference between SEIZ-TE and SEIZ-IS is less than 200 ms. Because it is possible that in the PSTN exchange a call related line test may be performed immediately before offering a call, there may be some time difference between the registration of the seizure in the call handling process and the sending of SEIZ-IS on the line. Therefore, TE should give right of way to the terminating call when, within 500 ms after SEIZ-TE for an originating call, a SEIZ-IS for a terminating call is received; see the signalling diagram.
Telephone TE intended to be connected to an SLT, shall in general not be provided with means for detecting double seizure; double seizure will then result in an very quick answering of the terminating call.
In case the CLIP supplementary service is applied to the SLT, double seizure will also result in early answer of the terminating call; the transfer of CLI digits is then not possible.

Blocking

The PSTN exchange can block the line, e.g. because of technical or administrative reasons, (temporary) failures or maintenance activities; the DC feeding voltage is then removed from the line. Unblocking takes place by supplying the feeding voltage again.
For TE, no blocking signal is available in the signalling.

Locking out
The line can be locked out when the exchange detects an unjustly occupation of the line, i.e. the line is seized for an originating call and no dialling information is received in time, or the network has indicated that, at the end of a call, the connection in the network has been released, but no release signal has been received in time from the TE. The locked out condition may also appear when in the exchange a line failure has been detected.
In such locked out condition, the exchange will ensure that there is no needless occupation of network resources; the DC feeding voltage may be limited in order to restrict the loop current.
As soon as in this condition a release signal (idle condition) is received from the TE, the locked out condition will, eventually after some delay time, cease and the idle condition will be entered. Dependent on the type of exchange, the delay time may last from some seconds to several minutes.

6.2.3 Calling Line Identification Presentation (CLIP)

The detailed description about the supplementary service CLIP (Dutch: Nummerweergave) in this section is subdivided as follows:
1) General description;
2) CLIP procedure;
3) TE requirements for CLIP.

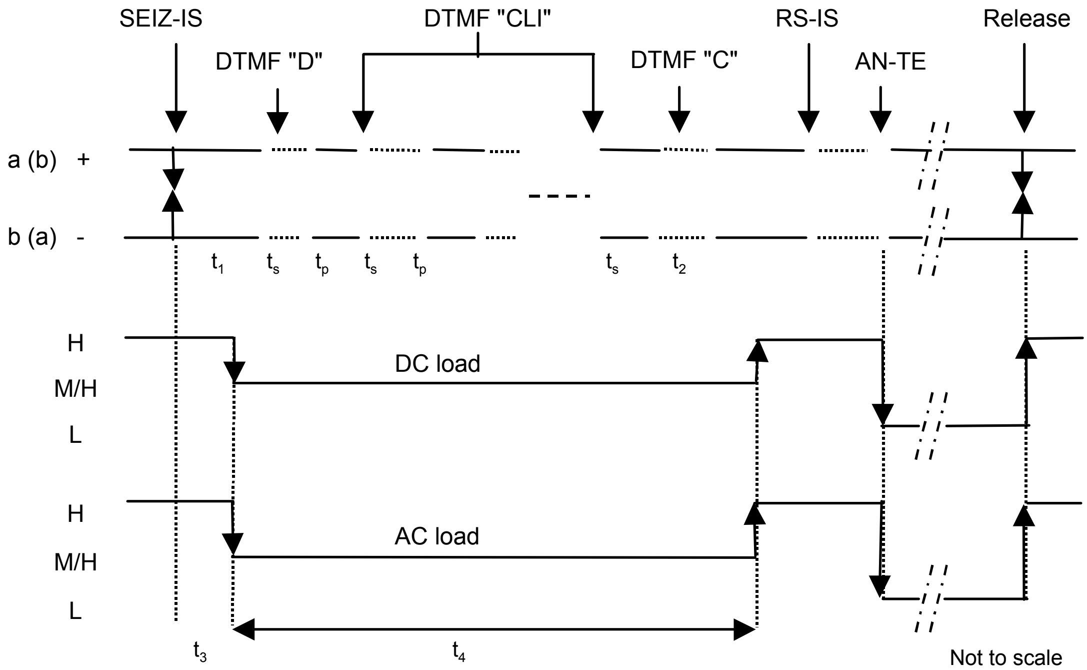
1) General description

CLIP provides the called party at an SLT with the possibility to receive during the terminating call handling procedure the calling party's telephone/ISDN number (CgPN: the identity of the calling party's NTP), provided that CLIP is activated for the called party's SLT and the calling party's number is available in the exchange and is allowed to be presented.
The number can be received and presented with the use of TE, which supports the CLIP function as provided by the PSTN of KPN Telecom; such TE is further indicated as CLIP-TE.
TE requirements for the support of CLIP are given hereafter (in the former national technical regulations, requirements were contained in T 11-12).
The CLIP service of KPN Telecom complies with ETSI EN 300 659-1, Annex B (normative) and ETSI EN 300 778-1, Annex A (normative).

Related to the CLIP service is the CLIR service. The CLIR supplementary service (Dutch: 'Blokkering Nummerweergave') enables the calling party to prevent the presentation of the customers telephone/ISDN number to the called party.
The default setting of CLIR is 'not activated', which means that presentation of the number is allowed; in agreement with the customer, CLIR can be activated permanently by KPN Telecom, which means that then presentation of the number is not allowed.
For the case of CLIR is not activated (meaning: presentation is allowed), the user can on a per call basis restrict the presentation by dialling the digit string: ∗31∗<CdPN> or 131<CdPN>. In this string, <CdPN> is the called party's telephone/ISDN number for establishing the call.

In the signalling procedure for the terminating call the CLI number information is transferred after the exchange has reversed the line feeding polarity (seizure signal from the exchange) and before the ringing signal is supplied to the line. The number information is transferred with DTMF signals. Only during this number transfer phase, specific requirements apply for CLIP-TE; outside this phase, the normal requirements apply. Related to the CWH supplementary service, it is not possible to present the CLI number of the second call, which has been arrived during the active phase of the first call.

In figure II-6/1, the CLIP procedure and the required behaviour of CLIP-TE is visualized; the case, in which the call has been answered by the CLIP-TE, is presented.



300 ms < t_1 < 800 ms   0 s < t_2 < 1 s t_3 < 250 ms   4,5 s < t_4 < 5,0 s

t_s / t_p: sending time / pause time DTMF code signals

Other indications: see explanation subsection 6.2.5

Figure II-6/1 Procedure CLIP, call answered by CLIP-TE

2) CLIP procedure

The CLIP procedure is applicable for SLTs for which the CLIP service is activated.
The following phases in the CLIP procedure can be identified:
a) Seizure phase of the line for a terminating call with CLIP;
b) CLI Number Information Transfer (NIT) phase;
c) Ringing/answering phase.

a) Seizure phase

As soon as a call has to be offered to an SLT for which CLIP is activated, the exchange shall seize the line for a terminating call. From that moment on, a lowohmic loop termination (≤ 560 Ω) at the NTP shall be interpreted

as an answer signal (one of the TEs in the subscriber's premises installation may have entered the loop state); therefore, if a lowohmic loop is detected during the CLIP procedure, the CLIP procedure shall be ceased and the call answered state is entered.
The seizure of the line is signalled to the TE by reversing the polarity of the feeding voltage (SEIZ-IS).
The idle (I) polarity of the feeding voltage of an SLT is not uniquely defined; the I- polarity may be 'a-wire positive or negative with respect to b-wire'.

b) CLI Number Information Transfer (NIT) phase
Between 300 and 800 ms after SEIZ-IS, the exchange shall start the transfer of the number information by sending a series of DTMF code signals. For a reliable transfer of the DTMF signals, it is necessary that the loop current is at least 0,4 mA.
The DTMF signals are transferred according to the DTMF standard ETSI ES 201 235, parts 1, 2 and 3 (see section 4.6.5); the DTMF transmitter is located in the PSTN exchange and the DTMF receiver in the TE. The sending time t_s of a DTMF code signal and the pause time t_p between two DTMF code signals are both > 65 ms; in general, the sending time per digit shall be: $t_s + t_p < 250$ ms.

The series DTMF code signals is composed of:
<D><CgPN><C>

The meaning is as follows:

<D>:	DTMF code signal "D" (start code "D")
<CgPN>:	Telephone/ISDN number of the calling NTP (calling line)
<C>:	DTMF code signal "C" (stop code "C")

<CgPN> is a series of DTMF code signals for the digits 0 up to and including 9, which represents a number of maximum 17 digits.
The following possibilities exist:
- In case of a national originated call: <CgPN> is composed of the national (trunk) prefix '0' followed by the national (significant) number;
- In case of an international originated call: <CgPN> is composed of the international prefix '00' followed by the country code and the national (significant) number in the originating country;
- In case the calling party's NTP identity is not available in the exchange, or is available, but presentation is not allowed due to the CLIR service: for <CgPN>, 10 times the digit "0" is sent.

If during the NIT phase an AN-TE is detected (e.g. one of the TEs connected to the NTP has changed to the lowohmic loop state), the exchange shall cease the sending of number information and the call answered state shall be entered.

c) Ringing/answering phase
In the normal CLIP procedure, the exchange shall between 0 and 1 s after the stop code "C" supply the Ringing Signal (RS) to the line; from that moment on, the normal call handling procedure is applicable.

3) TE requirements for CLIP

For a reliable transfer and reception of the number information, the CLIP-TE shall present specific physical and electrical conditions to the line; this state of the TE is indicated as Number Information Transfer (NIT) state.

Starting from the quiescent state of the TE and feeding voltage with idle polarity, as applicable for the line in idle condition, the NIT state shall be established within 200 ms after the reversal of the feeding voltage polarity and the reversed voltage has reached a level of higher than 30 V_{DC} (SEIZ-IS). Other polarity reversals as applied in the signalling, i.e. from I-polarity to R-polarity as NR-IS for an originating call and from R-polarity to I-polarity as R-IS (see following sections 6.3 and 6.4), should not result in establishing the NIT state.
It should be noted that feeding interruptions as applied during tests are no polarity reversals.

In the NIT state, the access requirements apply as for the quiescent state, except for those requirements as stated in the following:

- The DC resistance shall be between 90 kΩ and 110 kΩ. This requirement allows parallel connection of up to 5 CLIP-TEs at the NTP. If the CLIP-TE is not intended to be connected in parallel with other CLIP-TE, the requirement value is between 20 kΩ and 110 kΩ.
- The AC impedance in the voice band shall be greater than 1800 Ω, and preferably lower than 2400 Ω. This requirement allows parallel connection of some CLIP-TEs. If the CLIP-TE is not intended to be connected in parallel with other CLIP-TE, the value of the AC impedance may be such that, referenced to 600 Ω or to the ETSI reference impedance Z_R, the return loss is not less than 8 dB.

The requirements for reception of DTMF signals are in accordance with ES 201 235-3, taking into account for the detection levels that up to 5 CLIP-TEs can be connected in parallel.
For CLIP-TE with an AC impedance between 1800 Ω and 2400 Ω, the following detection levels may be applied:

- a level > 37 mV shall be recognized as a valid DTMF signal;
- a level < 28 mV shall not be recognized as a valid DTMF signal.

The detection times are specified as follows:

- a valid DTMF signal > 40 ms shall be interpreted as a valid digit;
- a valid DTMF signal < 20 ms shall not be interpreted as a valid digit.

The pause times between DTMF signals are specified as follows:

- If no valid DTMF signal is present during > 40 ms, this shall be interpreted as an interdigit pause;
- If no valid DTMF signal is present during < 20 ms, this shall not be interpreted as an interdigit pause.

When the CLIP information transfer is completed, but ultimately 5 s after the entering of the NIT state, the CLIP-TE shall leave the NIT state and return to the quiescent state with the ringing function, if provided, enabled, unless the TE has already left the NIT state while being forced to the loop state (i.e. by answering the call).
The CLIP information transfer can be regarded as completed when one of the following criteria are met:

a) The DTMF code "C" (stop code) is received;
b) Ringing signal is received;
c) The polarity of the feeding voltage is reverted back to the applicable I-polarity;
d) No DTMF code is received within 1 s after the reversal of the polarity of the feeding voltage from I to R and the reversed voltage has reached a level of higher than 30 V_{DC};
e) After receipt of a DTMF code, the DTMF pause is present for more than 1 s.

At least the criteria d) and e) should be supported by the CLIP-TE, as these criteria will guarantee in both normal and abnormal CLIP procedures, that the NIT state is left before or as soon as possible after the line has changed to the loop state because the call is answered by one or another TE connected to the NTP.

CLIP-TE may be NIT-only TE with only the function of receiving and displaying the CLI number, e.g. intended to always being connected in parallel with other TE (with or without such function). For NIT-only TE, only the quiescent state and the NIT state are applicable. A NIT-only TE should contribute as less as possible to the loading of the NTP.
The load of a NIT-only TE can be neglected if the following requirements are met:

- Both in quiescent state and in NIT state the capacitance at 25 Hz is less than 30 nF and the absolute impedance is greater than 200 kΩ;
- The DC resistance in quiescent state is greater than 4 MΩ;
- The AC impedance in the voice band in quiescent state is greater than 50 kΩ.

6.2.4 Call Waiting Hookflash (CWH)

The CWH supplementary service (Dutch: 'Wisselgesprek') permits the customer with an SLT to have the network, during an originating or terminating call in 'active state', indicating a new incoming call to the user by sending a special tone, the 'call waiting tone'; this new call is then in the 'call indicating state'.
The user can, with the aid of the hookflash signal, instruct the exchange to park the first ('active') call in a 'call waiting state' and to answer the new call (change the state of the second call from 'indicating state' to 'active state'). Subsequently, the user can, with the hookflash signal, switch between one and the other call. There can be only one call in 'indicating state' or in 'waiting state'.
Hookflash is a short highohmic pulse (> 17 kΩ for 90 - 800 ms). It should be noted that from a TE with a Register Recall (RR) button, with which a loop interruption (RR-signal) between 90 and 130 ms is generated, the HF signal can be sent by pressing the RR button.
If a loop interruption of longer than 1 s is received in the exchange, then this is recognized as a release indication and both the 'active' and 'waiting' call are released.

CWH can be activated permanently by KPN Telecom in agreement with the customer.
The customer with CWH activated, can deactivate CWH on a per call basis; if the customer wants to originate a call, during which no indication

of a new call is wanted, CWH can be deactivated for the duration of that originating call by dialling the digit string: #43∗<CdPN>.
In this string, <CdPN> is the called party's telephone/ISDN number for establishing the call.

6.2.5 Explanation of used indications and abbreviations

IS	Infrastructure
TE	Terminal Equipment
O	Originating call
T	Terminating call
CCBS	Completion of calls to busy subscriber
CLIP	Calling line identification presentation
CWH	Call waiting hookflash
P-IS	Polarity of the DC feeding voltage from the IS
I	Idle polarity; 'a-wire positive or negative with respect to b-wire'
R	Reversed polarity; 'reversed with respect to idle polarity'
∞	Feeding interruption
DC-TE	NTP presented DC loop termination of the TE installation
H	Highohmic; $\geq$ 400 kΩ
M/H	Medium Highohmic; 90 kΩ $\leq$ M/H $\leq$ 110 kΩ
L	Lowohmic; $\leq$ 560 Ω
RNR	Reference number of state/signal in signalling diagram
SEIZ	Seizure
RTR	Ready to receive
NR	Number received
NIT	Number information transfer with DTMF for CLIP
AN	Answer
R-IS	Release from IS
R-TE	Release from TE
HF	Hookflash for CWH; DC-TE: > 17 kΩ
BLK	Blocking
xx-IS	Signal sent from IS
xx-TE	Signal sent from TE
DT	Dial tone
RS	Ringing signal
RT	Ringing tone
RLT	Release tone
DTMF	DTMF dialling
DP	Decadic pulse dialling
MP	Metering pulse

s	$\geq$ s ms	Sending time of continuous signal: The signal shall be send without interruptions during at least s ms.
s	s_1 - s_2 ms	Sending time of pulsed signal: The signal shall be send without interruptions during a time between s_1 and s_2 ms.
r	r_1 - r_2 ms	Recognition time of signals; the following cases are identified: - A continuous signal: The signal shall be recognised such that a duration < r_1 ms is not seen as a valid signal, and a duration > r_2 ms is always seen as a valid signal. This also applies for recognition of dialling pulses. - A pulsed signal: Only a signal with a duration between r_1 and r_2 ms shall be recognised as a valid signal.

6.3 Call from TE to PSTN

Call type O: Originating call

6.3.1 Idle

In the idle condition of the line, the polarity of the feeding voltage is I and the loop termination in the TE is highohmic. The line feeding voltage in idle condition may be supplied from a source with a higher internal DC impedance than the source which is used in the line occupied condition; see section 4.6.

An earlier originating call to a busy destination, for which CCBS has been activated (see subsection 6.3.3), can be completed as soon as that destination has become free, by offering a CCBS recall in this idle state of the line.
The procedure for the CCBS recall is as follows:
The line is seized by the exchange by reversing the polarity of the line feeding voltage to R-polarity. At the same time, Ringing Signal superimposed on the feeding voltage is supplied to the line for maximum 20 s. The characteristics of the Ringing Signal for the CCBS recall is the same as for a normal terminating call.

- If the CCBS recall is answered by the caller within these 20 s, then the call set up to the (original dialled) destination is started; for the call handling procedure, the same condition as for the Number Received state for a normal call is reached and further procedures are as for a normal originating call; see subsection 6.3.3. In some type of PSTN exchanges it is possible that, during the transition from the 'Recall ringing state' to the 'Number Received state', the polarity of the line feeding voltage is reverted back to Idle polarity for a short time.
- If the CCBS recall is not answered in time by the caller, then the exchange shall bring the line back to the idle state (release the line) by reverting back the polarity of the feeding voltage to I-polarity.

6.3.2 Seizing from TE (SEIZ-TE)

For an originating call, the line is seized by the TE by making the loop termination lowohmic (SEIZ-TE).
In case of a both-way line, it is possible that the originating call collides with a terminating call (double seizure; see foregoing subsection 6.2.2). The lowohmic loop termination of the TE for the originating call is then interpreted in the exchange as a quick answer of the terminating call; the call handling and signalling procedure has to be changed to the related state in the terminating call process (call type T; section 6.4 hereafter).

6.3.3 Dialling from TE

After detection of SEIZ-TE, the exchange has to be prepared for the reception of dialling information. If in the related exchange the feeding voltage in idle condition is supplied from a more highohmic source than in the occupied condition, then a switch over to the feeding arrangement for the occupied condition has to take place. Further, it is possible that in this phase a pre call test on the line is performed in the exchange, which might also be noticeable as a short feeding interruption.

As soon as the exchange is Ready To Receive (RTR-IS) dialling information, dial tone (DT-IS) is sent; the voltage polarity remains to be the I-polarity.
The dialling information can be the called party number for an originating call or, only in case of DTMF dialling, a control command for a supplementary service.
The dialling information (digits) can be sent in with DTMF dialling (DTMF-TE) or with Decadic Pulse dialling (DP-TE); for these dialling systems, see section 4.6.
At detection of a dialling information signal, the dial tone is switched off.

After having received all dialling information, the exchange sends the number received signal (NR-IS) by reversing the polarity of the feeding voltage to R-polarity. NR-IS is also sent in cases that sending of digits makes no sense anymore because the exchange has concluded that the call can not be completed successfully, e.g. network congestion, not allowed destination, time out of the digit entering guard, faults, and such.
Acoustical indications following the reception of dialling information, both positive and negative indications (see section 3.3.4), are always provided in the NR-IS state (polarity R).

In the case that all dialling information has been received and the call has been routed to the called destination, but this destination is detected to be busy, the network does provide busy tone to the caller. During the time that busy tone is present and if CCBS is applicable for the called destination, the caller can activate the CCBS supplementary service by sending the digit <**5**>, either with DTMF dialling or with Decadic Pulse dialling, and then releasing the line. It is not possible on a PSTN access to provide information about the applicability of CCBS at the called destination. For information on CCBS, see the description of CCBS in chapter 2.

When time out of the time guard for listening to the acoustical indication occurs, the exchange shall send release (R-IS) with release tone; see subsection 6.3.7.
In case of time out of the digit entering guard, it is possible that NR-IS (without tone) is sent for 100 - 400 ms, followed by R-IS with release tone (RLT: same characteristic as congestion tone); see subsection 6.3.7.

6.3.4 Answering

The analogue signalling for SLT is not provided with the possibility to transfer an answer signal to the originating TE; for call type O, the DC loop conditions in the states NR-IS and "AN-IS" are the same. As soon as a call with a positive indication (Ringing tone (RT)) is answered, the sending of RT is stopped.
In the case that the call is not a free of charge call and the SLT is provided with the metering facility, the first metering pulse is sent at answer instant and can be used as an answer indication.

Note:
The 50 Hz 'common mode' metering facility is no longer commercially available. The present version of the publication does contain the technical information about the facility, because the facility is still present in the network

6.3.5 Highohmic loop termination from TE in NR/"AN" state

The change of the loop termination in the TE from lowohmic (NR/"AN" state) to a highohmic one is recognized in the exchange, in which the following situations are distinguished:

- If the duration of the highohmic condition is longer than 1000 ms, then the signal is interpreted as release signal (R-TE).
- If the duration of the highohmic condition is shorter than 1000 ms followed by a lowohmic loop for at least 150 ms, then the reaction is dependent of the SLT:
  a) An SLT without the CWH supplementary service:
     The signal may be interpreted as R-TE or may be ignored in which case the call remains active.
  b) An SLT with the CWH supplementary service:
     The signal can be interpreted as hookflash signal (HF-TE), with the understanding that it is only sure that a received highohmic pulse with a duration between 80 and 850 ms shall be interpreted as HF signal. If, according to CWH (see foregoing subsection 6.2.4), a second call (in 'call indicating' or 'call waiting' state) for the SLT is present, then the HF signal causes the switch over of the active call condition from one call to the other. If HF is received, but no second call is present, the HF signal shall be ignored and the existing call remains active.

6.3.6 Release initiated from TE (R-TE before R-IS)

Release of the call from the TE is initiated by a highohmic loop termination for longer than 1000 ms (R-TE).
If the call is in a state before NR-IS, i.e. the idle polarity I is present, then the idle state is reached with the detection of R-TE; a short feeding interruption due to switching over of feeding voltage supply or to a test may be noticed.
If the call is in the NR/"AN" state, i.e. the polarity is R, and the exchange has interpreted the signal as R-TE (see subsection 6.3.5), then R-IS is sent by reverting back to the voltage polarity I; the idle condition is then reached.

6.3.7 Release initiated from IS (R-IS before R-TE)

Release of the call from the IS is initiated by reverting the polarity of the feeding voltage from R to I (R-IS). If the exchange has to release the call in a phase that the polarity is not yet R (i.e. before the NR state), then the NR state with polarity R is first provided (eventually during a short time) in order to make the sending of R-IS possible; see also subsection 6.3.3.
In the R-IS state and before R-TE is received, the exchange shall also send release tone (RLT; same characteristic as congestion tone). In case of time out of the listening time guard to RLT, the sending of RLT is stopped and the line may be locked out.
After the TE (or the user) has recognised R-IS (polarity I with RLT), the line is set in the idle condition by providing during at least 1000 ms the highohmic loop termination from the TE.
A short feeding interruption due to switching over of feeding voltage supply or to a test may be noticed.

6.3.8 State transition and signalling diagrams, originating call

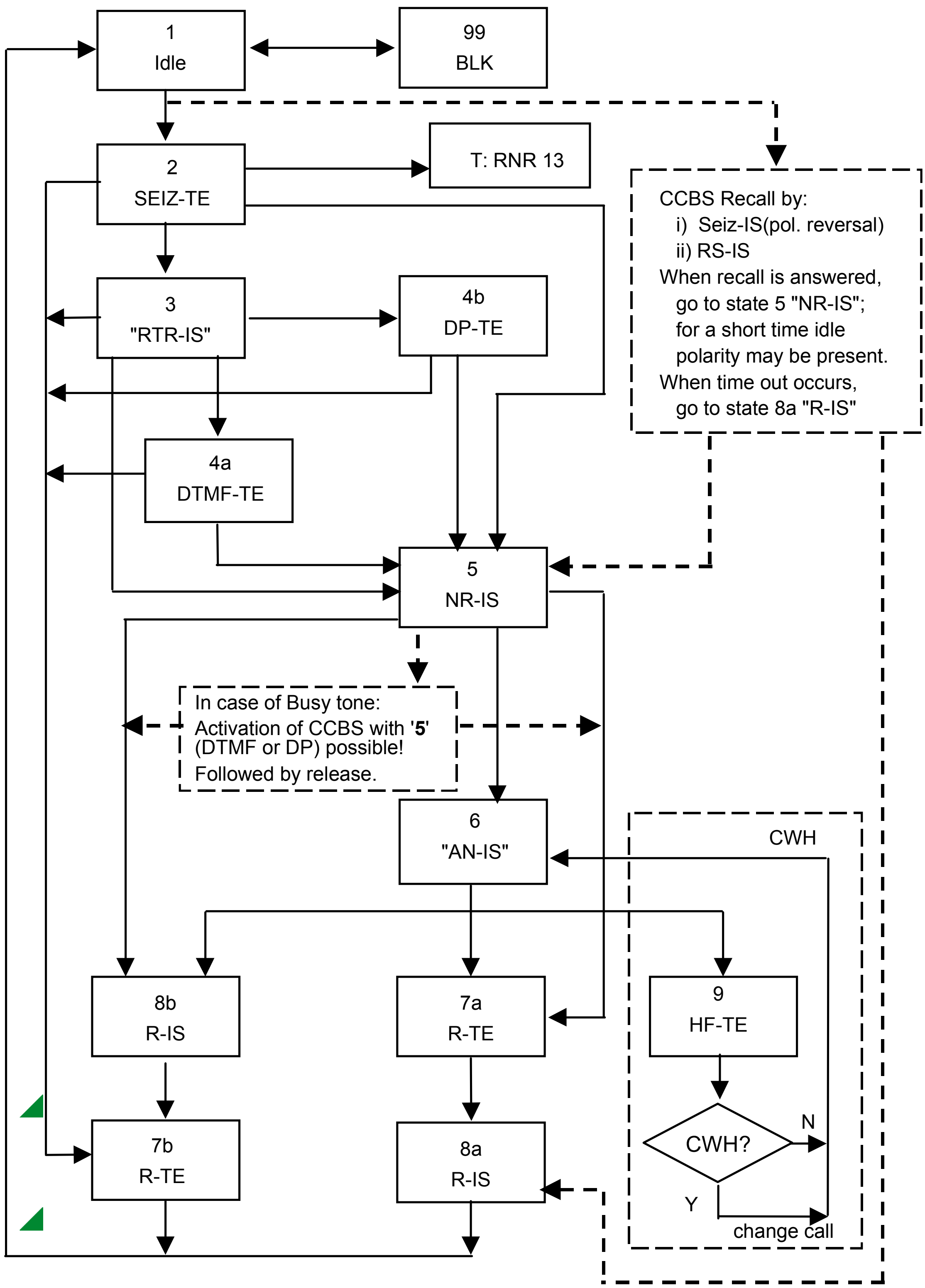


Figure II-6/2 State transition diagram, call type O

Table II-6/1 Signalling diagram, call type O

RNR	State/signal	P-IS		DC-TE	Sending/recognition time, other conditions
99	BLK From 1	∞	→	H	Blocking by switching off the DC line voltage. s: continuous, > 1 s
1	Idle After 7, 8a, 99	I	→ ←	H	Eventually unblocking by switching on the DC line voltage. *For CCBS Recall, see subsection 6.3.1*
2	SEIZ-TE After 1 (Possibly DC line voltage from a highohmic source and/or feeding interruption due to a test)	I	←	L	s: continuous; r: 10 – 200 ms On both-way lines: during 500 ms after SEIZ-TE a SEIZ-IS may be received (call collision). In that case, the terminating call takes priority: Change call type: → T: RNR 13.
3	DT-IS ("RTR-IS") After 2	I DT	 →	L	Dial tone as RTR-IS signal. Expiration of DT send guard: → RNR 5.
4a	DTMF-TE After 3	I	 ← . ←	L DTMF	DTMF code: s: >65 ms r: 20 - 40 ms DTMF pause: s: >65 ms r: 20 - 40 ms Digit entering guard: 6 – 8 s. Expiration of digit entering guard: → RNR 5.
4b	DP-TE After 3	I	← . . . . ←	L H L . . L	Break pulse: s: 50 – 72 ms r: 10 - 30 ms Make pulse: s: 31 – 46 ms r: 8 - 24 ms Digit pause: s: >400 ms r: 100 - 200 ms Digit entering guard: 6 – 8 s. Expiration of digit entering guard: → RNR 5.
5	NR-IS After 2, 3, 4a/b Acoustic / tone indication	R	→ Tone →	L	s: continuous, >150 ms; r: 10 – 100 ms Expiration of tone send guard: → RNR 8b. *In case of busy tone→CCBS possible, see subsection 6.3.3*
6	"AN-IS" After 5, 9	R	→ (MP)	L	At call answer, RT is stopped. If applicable, MPs are provided.
7a	R-TE After 5, 6	R	←	H	s: continuous, >1000 ms
8a	R-IS (after R-TE) After 7a	I	→	H	Idle state (to be maintained >500 ms).
8b	R-IS (before R-TE) After 5, 6	I RLT	→ →	L	s: continuous; r: 10 – 100 ms TE shall send R-TE within 6 s. Expiration of RLT send guard: line may be locked out.
7b	R-TE After 2, 3, 4a/b, 8b	I	←	H	Idle state. s: continuous, >1000 ms Possible feeding interruption due to a test
9	HF-TE After 6	R	 ←	L HF L	HF-pulse followed by L >150 ms. s: 90 - 800 ms r: 80 ms < HF-pulse < 850 ms CWH? No:→ ignore HF-pulse Yes:→ change call

6.4 Call from PSTN to TE

Call type T: Terminating call

6.4.1 Idle

In the idle condition of the line, the polarity of the feeding voltage is I and the loop termination in the TE is highohmic. The line feeding voltage in idle condition may be supplied from a source with a higher internal DC impedance than the source which is used in the line occupied condition; see section 4.6.
Before a SEIZ-IS signal is received, a short feeding interruption may be noticeable due to a pre call test.

6.4.2 Seizing from IS (SEIZ-IS)

For a terminating call, the line is seized by the exchange by reversing the polarity of the line feeding voltage to R-polarity (SEIZ-IS). Also Ringing Signal (RS; see section 4.6) superimposed on the feeding voltage is supplied, either at the same time as the polarity reversal or, if CLIP is applicable for the SLT, after the transfer of the CLI number information (see foregoing subsection 6.2.3).
In case of double seizure on both-way lines, the terminating call precedes and a quick answer of the terminating call occurs; see subsection 6.2.2.
It should be noted that the voltage peaks of the RS are higher than the level of the feeding voltage; both IS and TE shall cope with that phenomenon on the line.

6.4.3 Answering from TE (AN-TE)

Answering of the terminating call takes place by providing a lowohmic loop termination from the TE (AN-TE). As soon as the exchange has recognised AN-TE, BS is switched off and the communication path is through connected in the network.
Some aspects related to automatic answering TE are dealt with in the previous chapter.
The lower recognition time for AN-TE is 20 ms. If switching actions have to be performed in the TE, this should not result for longer than 20 ms in a line current with a value, which could be interpreted in the exchange as caused by a lowohmic loop.
Another point to note is the use of overvoltage arrestors in TE; TE with such circuitry might load the ringing signal in a asymmetric way and may thus create a resulting DC current on the line, which could be interpreted as an answer signal (false answering).

6.4.4 Highohmic loop termination from TE in AN state

The change of the loop termination in the TE from lowohmic (AN state) to a highohmic one is recognized in the exchange, in which the following situations are distinguished:

- If the duration of the highohmic condition is longer than 1000 ms, then the signal is interpreted as release signal (R-TE).
- If the duration of the highohmic condition is shorter than 1000 ms followed by a lowohmic loop for at least 150 ms, then the reaction is dependent of the SLT:

a) An SLT without the CWH supplementary service:
The signal may be interpreted as R-TE or may be ignored in which case the call remains active.
b) An SLT with the CWH supplementary service:
The signal can be interpreted as hookflash signal (HF-TE), with the understanding that it is only sure that a received highohmic pulse with a duration between 80 and 850 ms shall be interpreted as HF signal. If, according to CWH (see foregoing subsection 6.2.4), a second call (in 'call indicating' or 'call waiting' state) for the SLT is present, then the HF signal causes the switch over of the active call condition from one call to the other. If HF is received, but no second call is present, the HF signal shall be ignored and the existing call remains active.

6.4.5 Release initiated from TE (R-TE before R-IS)

Release of the call from the TE is initiated by a highohmic loop termination for longer than 1000 ms (R-TE); the polarity of the feeding voltage is R.
As soon as the exchange has recognized and interpreted the signal as R-TE (see subsection 6.4.4), R-IS is sent by reverting back to the voltage polarity I; the idle condition is then reached.

6.4.6 Release initiated from IS (R-IS before R-TE)

Release of the call from the IS is initiated by reverting the polarity of the feeding voltage from R to I (R-IS).
In the R-IS state and before R-TE is received, the exchange shall also send release tone (RLT; same characteristic as congestion tone). In case of time out of the listening time guard to RLT, the sending of RLT is stopped and the line may be locked out.
After the TE (or the user) has recognised R-IS (polarity I with RLT), the line is set in the idle condition by providing during at least 1000 ms the highohmic loop termination from the TE.
A short feeding interruption due to switching over of feeding voltage supply or to a test may be noticed.

6.4.7 State transition and signalling diagrams, terminating call

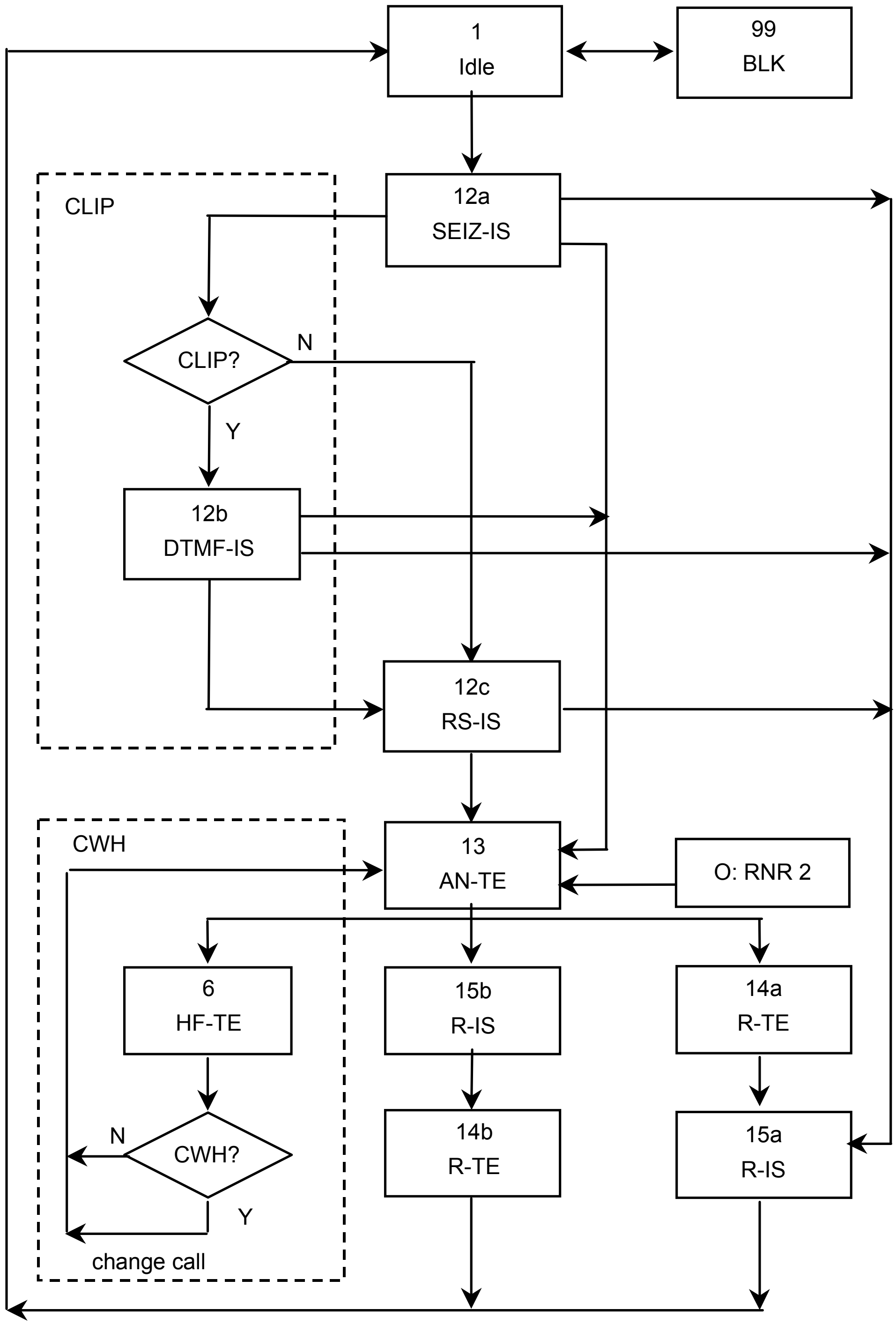


Figure II-6/3 State transition diagram, call type T

Table II-6/2 Signalling diagram, call type T

RNR	State/signal	P-IS		DC-TE	Sending/recognition time, other conditions
99	BLK From 1	∞	→	H	Blocking by switching off the DC line voltage. s: continuous, > 1 s
1	Idle After 14b, 15a, 99 (Possible feeding interruption due to switching action and/or test)	I	→ ←	H	Eventually unblocking by switching on the DC line voltage.
12a	SEIZ-IS After 1	R	→	H	s: continuous r: 10 - 60 ms CLIP? No:→ RNR 12c Yes:→ within 250 ms to RNR 12b
12b	DTMF-IS (If CLIP applies) After 12a	R DTMF R	← → . → ←	M/H H	DTMF code: s: >65 ms r: 20 - 40 ms DTMF pause: s: >65 ms r: 20 - 40 ms After 4,5 - 5,0 s: DC-TE shall be H again.
12c	RS-IS After 12a or 12b	R RS	 →	H	
13	AN-TE After 12a/b/c, 16 (possible after O: RNR2; see subsection 6.3.8)	R	←	L	s: continuous, >300 ms r: 20 - 200 ms; within 200 ms after start AN, switch off BS and switch through.
14a	R-TE After 13	R	←	H	s: continuous, >1000 ms
15a	R-IS After 14a (R-TE) and after 12a/b/c	I	→	H	Idle state (to be maintained >500 ms).
15b	R-IS (before R-TE) After 13	I RLT	→ →	L	s: continuous r: 10 - 100 ms TE shall send R-TE within 6 s. Expiration of RLT send guard: line may be locked out.
14b	R-TE After 15b	I	←	H	Idle state. s: continuous, >1000 ms Possible feeding interruption due to a test.
16	HF-TE After 13	R	 ←	L HF L	HF-pulse followed by L >150 ms. s: 90 - 800 ms r: 80 ms < HF-pulse < 850 ms CWH? No:→ ignore HF-pulse Yes:→ change call

7 Analogue signalling, PSTN multi-line termination

7.1 Introduction

This part II about the "Analogue PSTN network terminations" contains in chapter 3 a functional description of the basic PSTN call handling process, as this is perceivable on the subscriber line of the customer.
In sections 4.5 and 4.6, general aspects of subscriber line signalling, i.e. the physical and electrical appearance of the signals on the line, are dealt with.

In the present chapter, the detailed dynamic signalling procedures on the analogue subscriber line of a Multi-Line Termination (MLT) are described, including the procedures related to the Direct Dialling In (DDI) feature and to Malicious Call Identification (MCID).

The signalling is applicable on all MLTs.
TE (e.g. PBXs), intended to be connected to an MLT of the PSTN of KPN Telecom and complying at each of the line terminations with the former technical regulations, e.g. TBR 21, EN 301 437, TBR 38, and with the additional requirements in the present chapter, are expected to interwork correctly with the signalling.
The signalling as defined in the present chapter is in the first place specified for polarity dependent interworking with PBXs, but interworking with polarity independent TE is possible as well.
TE, connected to an MLT need not to be a PBX; e.g. it may also be a group of TEs to which calls are terminated from the PSTN by making use of the Group Number (GN) feature of an MLT.

The analogue MLT signalling is a DC loop signalling system for 2-wire analogue lines; the DC feeding voltage of the line is supplied by the PSTN exchange and the loop termination is applied by the TE. In general, use is made of continuous signals (line signalling states) corresponding with the actual phase of the call handling; for general information about other type of signals (e.g. pulsed signals, audio signals), see sections 4.5 and 4.6.

The continuous DC line signals from the exchange are provided in the form of the polarity of the DC feeding voltage; the polarity is further indicated as "Idle polarity (I)" and "Reversed polarity (R)"; I-polarity is 'a-wire positive with respect to b-wire', R-polarity is 'a-wire negative with respect to b-wire'.

Although the analogue signalling for MLT is provided with line condition signals from the IS to the TE by way of polarity reversals of the line feeding voltage, it is not necessary for TE to make use of these line signals. For polarity independent TE, the PSTN line signals are always accompanied by another indication, i.e. SEIZ-IS by Ringing Signal (RS) and R-IS before R-TE by release tone (RLT; same characteristic as congestion tone).

During switching actions in the exchange, short lasting feeding interruptions of up to 80 ms may occur, which have no signalling meaning; due to tests, the interruption time may be enlarged slightly.
The voltage level of a- and b-wire with respect to earth is dependent of the operative line circuit in the exchange; this level has no signalling meaning

During test and maintenance activities related to the line, special conditions, e.g. feeding interruptions, may be experienced which have no signalling meaning; information about these situations is provided in chapter 8.

The continuous line signals from the TE (or in fact from the subscriber's TE installation as presented at the NTP) are the 'very highohmic' loop termination (≥ 270 kΩ), the 'highohmic' loop termination (≥ 17 kΩ) and the 'lowohmic' loop termination (≤ 560 Ω). If in the highohmic condition in band information has to be transferred, i.e. Ringing tone (RT) or Busy tone (BT) from a PBX in case of a DDI call (see section 7.5), the loop termination shall be 'special highohmic' (between 17 and 80 kΩ).
The loop state detection function in the exchange is implemented such, that the at the NTP presented DC loop termination of the TE installation and the influence of the line are taken into account.

In following sections, the signalling procedures are described in relation with the different call types, i.e. an originating call (call type O), a terminating call without the DDI feature (call type T) and a terminating call with the DDI feature (call type D).
This chapter contains in section 7.2 a general introduction and the description of some special signalling conditions.
An explanation of the indications and abbreviations as used in this chapter can be found in subsection 7.2.3.

7.2 Introduction to the signalling procedures and special conditions

7.2.1 General

The detailed description of the signalling procedure for a call from TE to the PSTN, the Originating (O) call type, is provided in section 7.3.
For a call from PSTN to TE, the non-DDI and the DDI call type are distinguished; the non-DDI (the normal Terminating (T)) call type, is described in section 7.4 and the DDI (D) call type is described in section 7.5.
The procedure for the call type O and the call type T can respectively be compared with the originating and terminating call process in the basic PSTN call handling procedure; see chapter 3. Note that for the call type D, the terminating call handling procedure is extended with a dialling procedure from PSTN exchange to TE/PBX.
At the end of each section, the state transition diagram and the tabled signalling diagram for the related call type are provided.

In the signalling tables, the applicable parameter values of the different signalling states are indicated in a symbolic way; the direction of related signals are indicated with an arrow. Also information about the sending and recognition times and other conditions is provided.
By sending time of a continuous or pulsed signal is to be understood, the duration that the signal or the state has to be sent uninterrupted so that reliable recognition of the signal or state at the receiving side is possible.
Because the duration of signals may be increased or decreased while transmitting them over a long line, the sending time values differ from the recognition time values; the deviation between sending and recognition time values is necessary for a reliable interworking between PSTN and TE.

By recognition time of a state transition is to be understood: the duration that the new state has to be present before the new state is entered and actions related to the new state are started; herewith, it is considered that the new state is present from the time that the physical condition of the line conforms to the requirements of the new state.
Subsection 7.2.3 hereafter contains, beside an explanation of used indications and abbreviations, some further explanation about sending and recognition times.

7.2.2 Special signalling conditions

The following special signalling conditions are identified:
- Double seizure (call collision);
- Blocking;
- Locking out.

Double seizure
In case of both-way lines, it is possible that about the same time the TE sends the SEIZ-TE signal for originating a call and the exchange sends the SEIZ-IS signal for offering a terminating call; the SEIZ-TE signal and the SEIZ-IS signal "collide with each other".
In case of such double seizure, the terminating call from the PSTN has precedence; there is already a complete communication path through connected in the PSTN for this call.
The PSTN exchange has no means to detect double seizure. The implementation of the call handling in the PSTN exchange is such that a double seizure is only possible, when the time difference between SEIZ-TE and SEIZ-IS is less than 200 ms. Because it is possible that in the PSTN exchange a call related line test may be performed direct before a call, there may be some time difference between the registration of the seizure in the call handling process and the sending of SEIZ-IS on the line. Therefore, TE should give right of way to the terminating call when, within 500 ms after SEIZ-TE for an originating call, a SEIZ-IS for a terminating call is received; see the signalling diagram.
Polarity independent TE shall in general not be provided with means for detecting double seizure; double seizure will then result in a very quick answering of the terminating call.
Related to the subject of call collision on both way lines of an MLT, it is recommended to consult with KPN Telecom about the use of one of the available line hunting methods with the aim to decrease the chance on double seizure.

Blocking
The PSTN exchange can block the line, e.g. because of technical or administrative reasons, (temporary) failures or maintenance activities; the DC feeding voltage is then removed from the line. Unblocking takes place by supplying the feeding voltage again.
For TE, no blocking signal is available in the signalling.

Locking out
The line can be locked out when the exchange detects an unjustly occupation of the line, i.e. the line is seized for an originating call and no dialling information is received, or the network has indicated that, at the end of a call, the connection in the network has been released, but no release signal has been received in time from the TE. The locked out

condition may also appear when in the exchange a line failure has been detected.
In such locked out condition, the exchange will ensure that there is no needless occupation of network resources; the DC feeding voltage may be limited in order to restrict the loop current.
As soon as in this condition a release signal is received from the TE, the locked out condition will, eventually after some delay time, cease and the idle condition will be entered. Dependent on the type of exchange, the delay time may last from some seconds to several minutes.

7.2.3 Explanation of used indications and abbreviations

IS	Infrastructure
TE/PBX	Terminal Equipment / Private Branch Exchange
O	Originating call
T	Terminating call
D	Direct dialling in (DDI) call
P-IS	Polarity of the DC feeding voltage from the IS
a b + -	I = Idle polarity; 'a-wire positive with respect to b-wire'
a b - +	R = Reversed polarity; 'a-wire negative with respect to b-wire'
a b ∞	Feeding interruption
DC-TE	NTP presented DC loop termination of the TE installation
HH	Very highohmic; ≥ 270 kΩ
H	Highohmic; ≥ 17 kΩ
HS	Special highohmic; 17 kΩ ≤ HS ≤ 80 kΩ
L	Lowohmic; ≤ 560 Ω
RNR	Reference number of state/signal in signalling diagram
SEIZ	Seizure
RTR	Ready to receive
NR	Number received
AN	Answer
R-IS	Release from IS
R-TE	Release from TE
EIS	Establish idle state
MCID	Malicious call identification
BLK	Blocking
xx-IS	Signal sent from IS
xx-TE	Signal sent from TE
DT	Dial tone
RT	Ringing tone
RS	Ringing signal
BT	Busy tone
RLT	Release tone
DTMF	DTMF dialling
DP	Decadic pulse dialling
MP	Metering pulse

s	≥ s ms	Sending time of continuous signal: The signal shall be send without interruptions during at least s ms.
s	s_1 - s_2 ms	Sending time of pulsed signal: The signal shall be send without interruptions during a time between s_1 and s_2 ms.
r	r_1 - r_2 ms	Recognition time of signals: the following cases are identified:

- A continuous signal: The signal shall be recognised such that a duration < r_1 ms is not seen as a valid signal, and a duration > r_2 ms is always seen as a valid signal. This also applies for recognition of dialling pulses.
- A pulsed signal: Only a signal with a duration between r_1 and r_2 ms shall be recognised as a valid signal.

7.3 Call from TE to PSTN

Call type O: Originating call

7.3.1 Idle

In the idle condition of the line, the polarity of the feeding voltage is I and the loop termination in the TE is highohmic. The line feeding voltage in idle condition may be supplied from a source with a higher internal DC impedance than the source which is used in the line occupied condition; see section 4.6.

7.3.2 Seizing from TE (SEIZ-TE)

For an originating call, the line is seized by the TE by making the loop termination lowohmic for polarity I (SEIZ-TE).
In case of a both-way line, it is possible that the originating call collides with a terminating call (double seizure; see foregoing subsection 7.2.2). TE should recognize double seizure without causing an answer signal, e.g. by remaining for some time highohmic for polarity R. For polarity independent TE, the lowohmic loop termination of the TE for the originating call shall normally also be lowohmic for polarity R; this is then interpreted in the exchange as a quick answer of the terminating call.
In both cases, the call handling and signalling procedure has to be changed to the related state in the procedure for call type T (section 7.4) or call type D (section 7.5), dependent on whether the terminating call process is DDI or non-DDI for the MLT.

7.3.3 Dialling from TE

After detection of SEIZ-TE, the exchange has to be prepared for the reception of dialling information. If in the related exchange the feeding voltage in idle condition is supplied from a more highohmic source than in the occupied condition, then a switch over to the occupied condition feeding arrangement has to take place. Further, it is possible that in this phase a pre call test on the line is performed in the exchange, which might also be noticeable as a short feeding interruption.
As soon as the exchange is Ready To Receive (RTR-IS) dialling information, dial tone (DT-IS) is sent; the voltage polarity remains the I-polarity.

The dialling information is the called party number for the originating call; eventually, if and as far as user controlled supplementary services are provided to an MLT, control commands with DTMF signalling can be provided.
The dialling information (digits) can be sent in with DTMF dialling (DTMF-TE) or with Decadic Pulse dialling (DP-TE); for these dialling systems, see section 4.6.
At detection of a dialling information signal, the dial tone is switched off.

After having received all dialling information, the exchange sends the number received signal (NR-IS) by reversing the polarity of the feeding voltage to R-polarity. NR-IS is also sent in cases that sending of digits makes no sense anymore because the exchange has concluded that the

call can not be completed successfully, e.g. network congestion, not allowed destination, time out of the digit entering guard, faults, and such. Acoustical indications following the reception of dialling information, both positive and negative indications (see section 3.4), are always provided in the NR-IS state (polarity R).
When time out of the time guard for listening to the acoustical indication occurs, the exchange shall send release (R-IS) with release tone; see subsection 7.3.6. In case of time out of the digit entering guard, it is possible that NR-IS (without tone) is sent for 100 - 400 ms, followed by R-IS with release tone (RLT: same characteristic as congestion tone); see subsection 7.3.6.

7.3.4 Answering

The analogue signalling for MLT is not provided with the possibility to transfer an answer signal to the originating TE; for call type O, the DC loop conditions in the states NR-IS and "AN-IS" are the same. As soon as a call with a positive indication (Ringing tone (RT)) is answered, the sending of RT is stopped.
In the case that the call is not a free of charge call and the MLT is provided with the metering facility, the first metering pulse is sent at answer instant and can be used as an answer indication.

Note:
The 50 Hz 'common mode' metering facility is no longer commercially available. The present version of the publication does contain the technical information about the facility, because the facility is still present in the network

7.3.5 Release initiated from TE

Release before NR-IS
Before NR-IS, the I-polarity is still present on the line. Release from the TE is then provided by establishing the idle state from the TE (EIS-TE) with a very highohmic loop termination for polarity I during at least 500 ms.

Release in NR/"AN" state
In NR/"AN" state, the R-polarity is present on the line. Release of the call from the TE is initiated by a highohmic loop termination for longer than 500 ms (R-TE), either only for polarity R or independent of the polarity.
As soon as the exchange has interpreted this as R-TE, the exchange shall send R-IS by reverting back the polarity from R to I. A TE, which loop termination is not yet highohmic for I-polarity (e.g. in order to be able to detect R-IS), shall within 150 ms after the beginning of R-IS establish the idle condition (EIS-TE) by changing the loop termination to very highohmic for polarity R and maintaining this condition for at least 500 ms.

A short feeding interruption due to switching over of feeding voltage supply or to a test may be noticed.

7.3.6 Release initiated from IS (R-IS before R-TE)

Release of the call from the IS is initiated in NR/"AN" state (polarity R) by reverting the polarity of the feeding voltage back to I (R-IS). If the exchange has to release the call in a phase that the polarity is not yet R

(i.e. before the NR state), then the NR state with polarity R is first provided (eventually during a short time) in order to make the sending of R-IS possible; see also subsection 7.3.3.
R-IS before R-TE is always accompanied by release tone (RLT; same characteristic as congestion tone). In case of time out of the listening time guard to RLT, the sending of RLT is stopped and the line may be locked out.
After the TE (or the user) has recognized R-IS (polarity I with RLT), the TE shall establish the idle condition (EIS-TE) by providing during at least 500 ms the very highohmic loop termination for polarity I.
A short feeding interruption due to switching over of feeding voltage supply or to a test may be noticed.

7.3.7 State transition and signalling diagrams, originating call

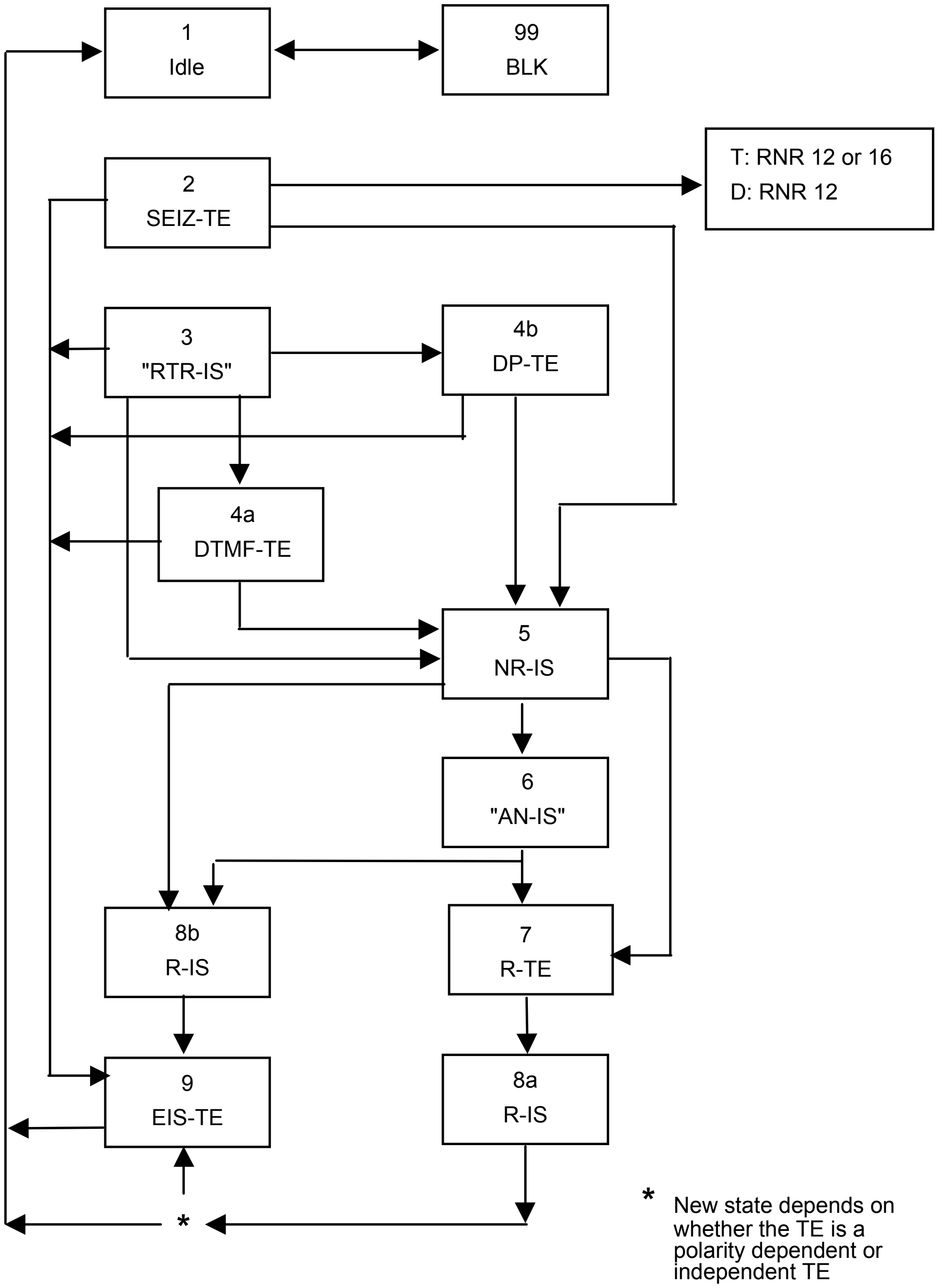


Figure II-7/1 State transition diagram, call type O

Table II-7/1 Signalling diagram, call type O

RNR	State/signal	P-IS a b		DC-TE	Sending/recognition time, other conditions
99	BLK From 1	∞	→	H	Blocking by switching off the DC line voltage s: continuous, > 1 s
1	Idle After 8a, 9, 99	+ -	→ ←	HH	Eventually unblocking by switching on the DC line voltage.
2	SEIZ-TE After 1 (Possibly DC line voltage from a highohmic source and/or feeding interruption due to a test)	+ -	←	L	L for I-polarity: s: continuous r: 10 – 200 ms On both-way lines: during 500 ms after SEIZ-TE a SEIZ-IS may be received (call collision). In that case, the terminating call takes priority. Change call type as appropriate: → T: RNR 12 or 16 (subsection 7.4.8) → D: RNR 12 (subsection 7.5.8) Note: For detecting call collision, DC-TE may for some time remain H for R-polarity.
3	DT-IS ("RTR-IS") After 2	+ - DT	 →	L	Dial tone as RTR-IS signal. Expiration of DT send guard: → RNR 5.
4a	DTMF-TE After 3	+ -	 ← . . ←	L DTMF	DTMF code: s: >65 ms r: 20 - 40 ms DTMF pause: s: >65 ms r: 20 - 40 ms Digit entering guard: 6 - 8 s. Expiration of digit entering guard: → RNR 5.
4b	DP-TE After 3	+ -	← . . . . ←	L HH L . . L	Break pulse: s: 50 - 72 ms r: 10 - 30 ms Make pulse: s: 31 - 46 ms r: 8 - 24 ms Digit pause: s: >400 ms r: 100 - 200 ms Digit entering guard: 6 - 8 s. Expiration of digit entering guard: → RNR 5.
5	NR-IS After 2, 3, 4a/b Acoustic / tone indication	- +	→ Tone →	L	S: continuous, >150 ms r: 10 - 100 ms Expiration of tone send guard: → RNR 8b.
6	"AN-IS" After 5	- +	→ (MP)	L	At call answer, RT is stopped. If applicable, MPs are provided.
7	R-TE After 5, 6	- +	←	H	H for R-polarity or independent of polarity. s: continuous, >500 ms r: longer than 200 to 300 ms
8a	R-IS (after R-TE) After 7	+ -	→	HH or L *)	HH for I-polarity:→ Idle state (>500 ms). L for I-polarity:→ TE shall send within 150 ms EIS-TE (RNR 9).
8b	R-IS (before R-TE) After 5, 6	+ - RLT	→ →	L	s: continuous r: 10 - 100 ms TE shall send R-TE within 6 s. Expiration of RLT send guard: line may be locked out.
9	EIS-TE After 2, 3, 4a/b, 8a/b	+ -	←	HH	Idle state. s: continuous, >500 ms Possible feeding interruption due to a test.

*) Dependent on polarity dependency of TE

7.4 Call from PSTN to TE without Direct Dialling In

Call type T: Terminating call

7.4.1 Idle

In the idle condition of the line, the polarity of the feeding voltage is I and the loop termination in the TE is highohmic. The line feeding voltage in idle condition may be supplied from a source with a higher internal DC impedance than the source which is used in the line occupied condition; see section 4.6.
Before a SEIZ-IS signal is received, a short feeding interruption may be noticeable due to a pre call test.

7.4.2 Seizing from IS (SEIZ-IS)

For a terminating call, the line is seized by the exchange by reversing the polarity of the line feeding voltage to R-polarity (SEIZ-IS). At the same time Ringing Signal (RS) superimposed on the feeding voltage with polarity R is supplied to the line. In TE, the polarity reversal and/or the RS can be used as seizure indication from the IS.
In case of double seizure on both-way lines, the terminating call precedes; see foregoing subsection 7.2.2. If the TE is not yet lowohmic for polarity R, the SEIZ-IS state remains; if the TE is also lowohmic for polarity R, the AN-TE state is entered (quick answer).
The voltage peaks of the RS are higher than the level of the feeding voltage; both IS and TE shall cope with that phenomenon on the line.
After detection of the line seizure from the IS, TE shall during the whole call handling process be prepared to receive release from the exchange (R-IS); see subsection 7.4.6.

7.4.3 Answering from TE (AN-TE)

Answering of the terminating call takes place by providing a lowohmic loop termination in the TE for polarity R (AN-TE). As soon as the exchange has recognised AN-TE, BS is switched off and the communication path is through connected in the network.
The lower recognition time for AN-TE is 20 ms. If switching actions have to be performed in the TE, this should not result for longer than 20 ms in a line current with a value, which could be interpreted in the exchange as caused by a lowohmic loop.
Another point to note is the use of overvoltage arrestors in TE; TE with such circuitry might load the ringing signal in a asymmetric way and may thus create a resulting DC current on the line, which could be interpreted as an answer signal (false answering).

7.4.4 Highohmic loop termination from TE in AN state

The change of the loop termination in the TE from lowohmic (AN state) to a highohmic one is recognized in the exchange, in which the following situations are distinguished:

- If the duration of the highohmic condition is longer than 400 - 500 ms, then the signal is interpreted as release signal (R-TE).
- If the duration of the highohmic condition is shorter than the recognition time for R-TE (400 - 500 ms), followed by a lowohmic loop for at least

150 ms, then, if MCID is activated for the MLT, the signal may be recognized as MCID pulse; else it will be ignored. Only a received highohmic pulse with a duration between 85 and 220 ms shall be interpreted as MCID-TE signal.

If not one of these conditions is met, the AN state is maintained.

7.4.5 Release initiated from TE (R-TE before R-IS)

In AN state, the R-polarity is present on the line. Release of the call from the TE is initiated by a highohmic loop termination for longer than 500 ms (R-TE), either only for polarity R or independent of the polarity.
As soon as the exchange has interpreted this as R-TE (see subsection 7.4.4), the exchange shall send R-IS by reverting back the polarity from R to I. A TE, which loop termination is not yet highohmic for I-polarity (e.g. in order to be able to detect R-IS), shall within 150 ms after the beginning of R-IS establish the idle condition (EIS-TE) by changing the loop termination to very highohmic for polarity I and maintaining this condition for at least 500 ms.
A short feeding interruption due to switching over of feeding voltage supply or to a test may be noticed.

7.4.6 Release initiated from IS (R-IS before R-TE)

Release of the call from IS in all signalling states with polarity R, is initiated by reverting the polarity of the feeding voltage back to I (R-IS).
R-IS before R-TE is always accompanied by release tone (RLT; same characteristic as congestion tone). In case of time out of the listening time guard to RLT, the sending of RLT is stopped and the line may be locked out.
After the TE (or the user) has recognized R-IS (polarity I with RLT), the TE shall establish the idle condition (EIS-TE) by providing during at least 500 ms the very highohmic loop termination for polarity I.
A short feeding interruption due to switching over of feeding voltage supply or to a test may be noticed.

7.4.7 State transition and signalling diagrams, terminating call

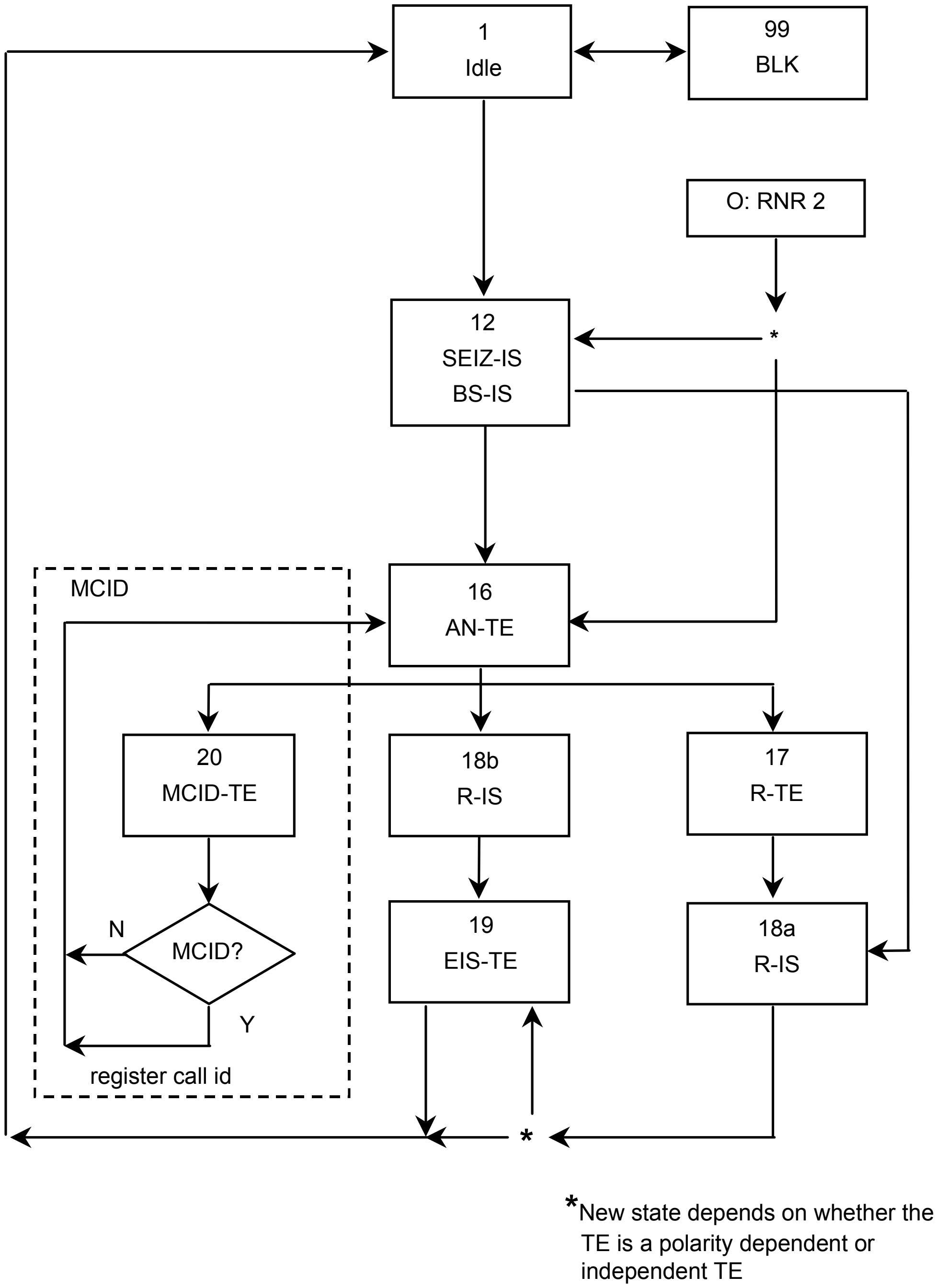


Figure II-7/2 State transition diagram, call type T

Table II-7/2 Signalling diagram, call type T

RNR	State/Signal	P-IS a b		DC-TE	Send / recognition time, other conditions
99	BLK From 1	∞	➔	H	Blocking by switching off the DC line voltage s: continuous, > 1 s
1	Idle After 14b, 15a, 99 (Possible feeding interruption due to switching action and/or test)	+ -	➔ ←	HH	Eventually unblocking by switching on the DC line voltage.
12	SEIZ-IS After 1; possibly also after O: RNR 2 *) (see subsection 7.3.7)	- +	➔	H	S: continuous; r: 10 - 60 ms Also sending of BS-IS.
16	AN-TE After 12, 20; possibly also after O: RNR2 *) (see subsection 7.3.7)	- +	←	L	S: continuous, >300 ms r: 20 - 200 ms; within 200 ms after start AN, switch off BS and switch through.
17	R-TE After 16	- +	←	H	H for R-polarity or independent of polarity. S: continuous, >500 ms r: longer than 400 to 500 ms
18a	R-IS After 17 (R-TE) and after 12	+ -	➔	HH or L *)	HH for I-polarity:➔ Idle state (>500 ms). L for I-polarity:➔ TE shall send within 150 ms EIS-TE (RNR 19).
18b	R-IS (before R-TE) After 16	+ - RLT	➔ ➔	L	S: continuous; r: 10 - 100 ms TE shall send R-TE within 6 s. Expiration of RLT send guard: line may be locked out.
19	EIS-TE After 18a/b	+ -		HH	Idle state. S: continuous, >500 ms Possible feeding interruption due to a test.
20	MCID-TE After 16	- +	 ←	L H L	H-pulse followed by L >150 ms S: 100 - 200 ms r: 85 ms < H-pulse < 220 ms MCID? No:➔ ignore H-pulse Yes:➔ register call id.

*) Dependent on polarity dependency of TE

7.5 Call from PSTN to TE with Direct Dialling In

Call type D: DDI call

7.5.1 Idle

In the idle condition of the line, the polarity of the feeding voltage is I and the loop termination in the TE/PBX is highohmic. The line feeding voltage in idle condition may be supplied from a source with a higher internal DC impedance than the source which is used in the line occupied condition; see section 4.6.
Before a SEIZ-IS signal is received, a short feeding interruption may be noticeable due to a pre call test.

7.5.2 Seizing from IS (SEIZ-IS)

For a DDI call, the line is seized by the exchange by reversing the polarity of the line feeding voltage to R-polarity (SEIZ-IS). No Ringing Signal (RS) is supplied to the line. Both TE and IS are then to be prepared for the transfer of dialling information from the IS to the TE for establishing a path through the PBX to a secondary connected TE.
In case of double seizure on both-way lines, the terminating DDI call precedes; see foregoing subsection 7.2.2. As the TE is still highohmic for polarity R, the SEIZ-IS state for the DDI call is applicable.
After detection of the line seizure from the IS, TE shall during the whole call handling process be prepared to receive release from the exchange (R-IS); see subsection 7.5.7.

7.5.3 Transfer of dialling information from IS to TE

As soon as the TE is ready to receive digits, but within 500 ms after SEIZ-IS, the TE shall indicate this by a lowohmic loop termination for polarity R (RTR-TE).
After detection of RTR-TE, the exchange shall send the DDI-digits using DTMF dialling; see section 4.6.
The lowohmic loop for polarity R shall be maintained continuously until the whole DDI number is received or further sending of digits does not make sense anymore. After having received all digits, the TE shall provide the special highohmic loop for polarity R as a number received indication (NR-TE) and shall then send Ringing tone (RT) or Busy tone (BT), dependent on the condition of the reached secondary line or TE.
On detection of NR-TE, the exchange shall switch through the transmission path to the calling party so that the provided RT or BT can be heard by the caller.

7.5.4 Answering from TE (AN-TE)

Answering of the terminating call takes place by providing a lowohmic loop termination in the TE for polarity R (AN-TE). As soon as the exchange has recognized AN-TE, the communication path is through connected in the network.
The lower recognition time for AN-TE is 20 ms. If switching actions have to be performed in the TE, this should not result for longer than 20 ms in a line current with a value, which could be interpreted in the exchange as caused by a lowohmic loop.

7.5.5 Highohmic loop termination from TE in AN state

The change of the loop termination in the TE from lowohmic (AN state) to a highohmic one is recognized in the exchange, in which the following situations are distinguished:

- If the duration of the highohmic condition is longer than 400 - 500 ms, then the signal is interpreted as release signal (R-TE).
- If the duration of the highohmic condition is shorter than the recognition time for R-TE (400 - 500 ms) followed by a lowohmic loop for at least 150 ms, then, if MCID is activated for the MLT, the signal may be recognized as MCID pulse; else it will be ignored. Only a received highohmic pulse with a duration between 85 and 220 ms shall be interpreted as MCID-TE signal.

If not one of these conditions is met, the AN state is maintained.

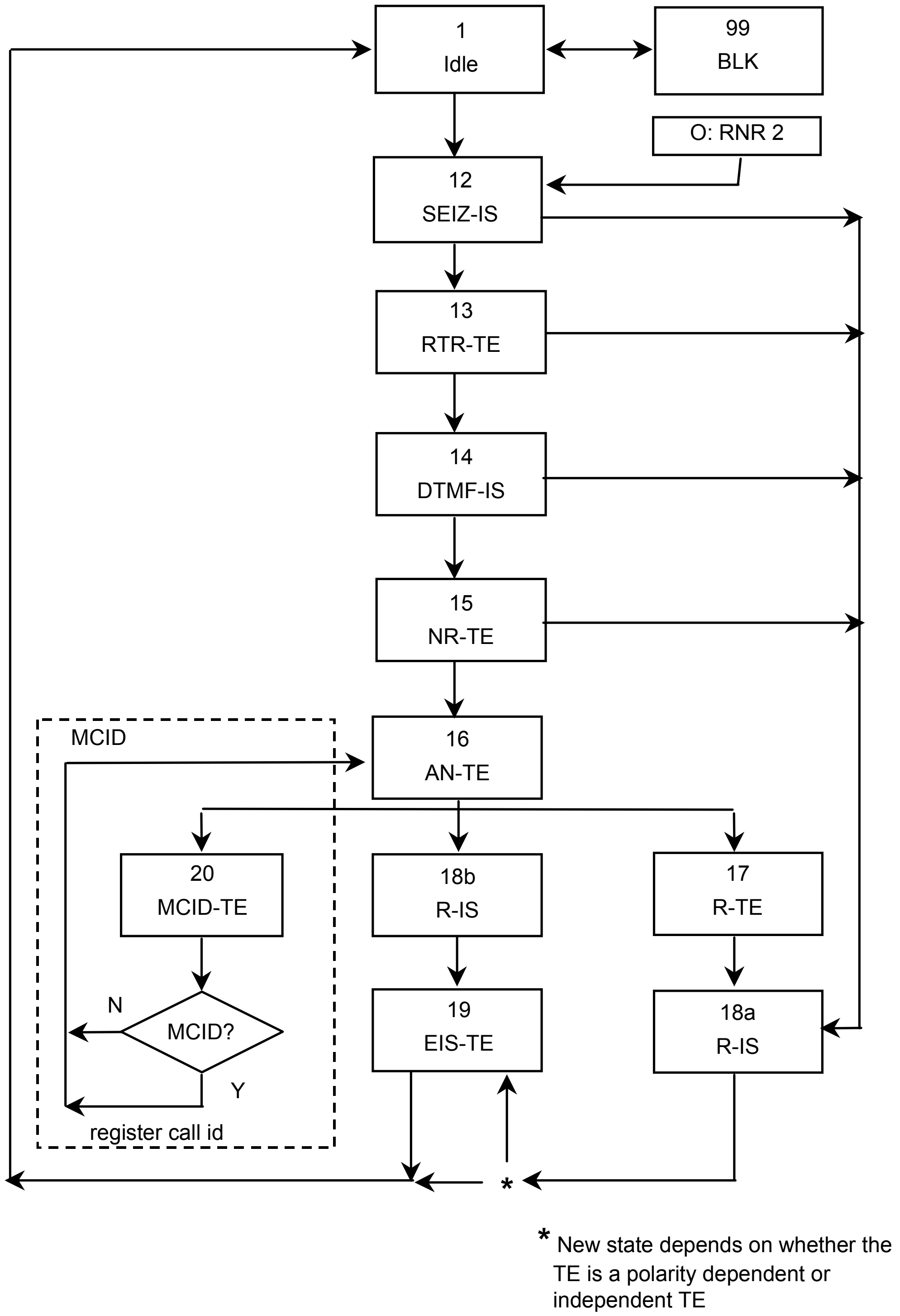
7.5.6 Release initiated from TE (R-TE before R-IS)

In AN state, the R-polarity is present on the line. Release of the call from the TE is initiated by a highohmic loop termination for longer than 500 ms (R-TE), either only for polarity R or independent of the polarity.
As soon as the exchange has interpreted this as R-TE (see subsection 7.5.5), the exchange shall send R-IS by reverting back the polarity from R to I. A TE, which loop termination is not yet highohmic for I-polarity (e.g. in order to be able to detect R-IS), shall within 150 ms after the beginning of R-IS establish the idle condition (EIS-TE) by changing the loop termination to very highohmic for polarity I and maintaining this condition for at least 500 ms.
A short feeding interruption due to switching over of feeding voltage supply or to a test may be noticed.

7.5.7 Release initiated from IS (R-IS before R-TE)

Release of the call from IS in all signalling states with polarity R, is initiated by reverting the polarity of the feeding voltage back to I (R-IS).
R-IS before R-TE is always accompanied by release tone (RLT; same characteristic as congestion tone). In case of time out of the listening time guard to RLT, the sending of RLT is stopped and the line may be locked out.
After the TE (or the user) has recognized R-IS (polarity I with RLT), the TE shall establish the idle condition (EIS-TE) by providing during at least 500 ms the very highohmic loop termination for polarity I.
A short feeding interruption due to switching over of feeding voltage supply or to a test may be noticed.

7.5.8 State transition and signalling diagrams, DDI call



FigureII-7/3 State transition diagram, call type D

Table II-7/3 Signalling diagram, call type D

PNR	State/Signal	P-IS a b		DC-TE	Send / recognition time, other conditions
99	BLK From 1	∞	→	H	Blocking by switching off the DC line voltage s: continuous, > 1 s
1	Idle After 18a, 19, 99 (Possible feeding interruption due to switching action and/or test)	+ -	→ ←	HH	Eventually unblocking by switching on the DC line voltage.
12	SEIZ-IS After 1; possibly also after O: RNR 2 (see subsection 7.3.7)	- +	→	H	s: continuous, r: 10 - 60 ms Sending of RTR-TE within 500 ms.
13	RTR-TE After 12	- +	→	L	s: continuous, > 150 ms, r: 20 - 100 ms The first digit is sent between 20 ms and 8 s.
14		- + DTMF	→ . . →	L	DTMF code: s: >65 ms, r: 20 - 40 ms DTMF pause: s: 65 – 8000 ms, r: 20 - 40 ms (the DTMF pause shall be close to the minimum as long as digits are available).
15	NR-TE After 14 (Also sending RT or BT from the TE)	- +	← ←	HS RT or BT	s: continuous, >300 ms, r: 10 - 100 ms NR-TE is to be sent within 300 ms after having received the last digit. The transmission path to the caller will be through connected.
16	AN-TE After 15, 20	- +	←	L	s: continuous, >300 ms r: 20 - 100 ms; through connection within 100 ms after start ANS.
17	R-TE After 16	- +	←	H	H for R-polarity or independent of polarity. s: continuous, >500 ms r: longer than 400 to 500 ms
18a	R-IS After 17 (R-TE) and after 12, 13, 14, 15	+ -	→	HH or L *)	HH for I-polarity: → Idle state (>500 ms). L for I-polarity: → TE shall send within 150 ms EIS-TE (RNR 19).
18b	R-IS (before R-TE) After 16	+ - RLT	→ →	L	s: continuous r: 10 - 100 ms TE shall send R-TE within 6 s. Expiration of RLT send guard: line may be locked out.
19	EIS-TE After 18a/b	+ -	←	HH	Idle state. s: continuous, >500 ms Possible feeding interruption due to a test.
20	MCID-TE After 16	- +	←	L H L	H-pulse followed by L >150 ms. s: 100 - 200 ms r: 85 ms < H-pulse < 220 ms MCID? No: → ignore H-pulse Yes: → register call id.

*) Dependent on polarity dependency of TE

8 Test and measurement conditions, PSTN network terminations

8.1 Introduction

In order to maintain a high grade of quality in the telecommunication network of KPN Telecom, different types of tests and measurements are performed on a regular basis.

Related to the network terminations, the following tests can be identified:

- Automated routine tests/measurements for the benefit of directing preventive and corrective maintenance actions;
- Tests in direct relation with a telephone call;
- Manual measurements, mostly as a consequence of detected faults or received complaints.

After a short description of these three measurement principles, information is provided about the physical phenomena on the line.

Manual measurements for corrective actions may interfere with terminal equipment. Automated tests are designed such that, in normal circumstances, these do not interfere with TE, which conforms to the former type approval requirements (see chapter 5) and is not sensitive for phenomena like short feeding interruptions.
For details about the physical phenomena on the line, see section 8.3.

8.2 The different measurement principles

8.2.1 Measurements for directing maintenance actions

For the benefit of maintaining a high grade of quality of the access network of the PSTN, automated measurements are performed on a regular basis. As a result of such test, it is possible that during some seconds voltage variations occur on the line; the voltage can be higher or can be switched off. Connected TE should not react to these variations.
It is the intention of KPN Telecom to minimize the hindrance, which users might experience during the tests. Automated routine tests are only performed on subscriber lines in idle condition; an active call shall never be interrupted.
A routine test may last for 1 minute as a maximum; for most lines it shall last for 15 seconds at the utmost and in average less than 10 seconds. Further, the tests are normally performed during night because the call intensity is then low.

8.2.2 Call related tests

Some type of exchanges perform tests on the line immediately before or after a call. Such pre call and after call tests are performed in order to check whether the line or network termination is able to handle the call or has handled the call correctly. Under circumstances this might be audible. The call related tests last for two seconds at the utmost.

8.2.3 Manual measurements and tests

During manual measurements, which are performed during repair activities or in case of extending and updating the access capability of exchanges, the user may experience some hindrance. This hindrance may be caused by short interruptions of the connection with the exchange or some interference with terminal equipment.

8.3 Phenomena at the network termination during tests

Tests and measurements are performed such that the interference with the signalling on the line is as less as possible. It is necessary that, during the measurements, the normal line voltage is switched off and test voltages are supplied to the line.

8.3.1 Deviation of the normal line voltage

During tests and measurements, DC line voltages with values of up to 100 V may be present during some seconds. These test voltages, are supplied from a source with such a high impedance that, with a short circuited loop, the maximum line current shall be less than 80 mA. The test voltage shall, in case of an unexpected change in the loop state of the TE, not cause any damage to the TE.
The test voltage may be supplied between the a- and b-wire as well as between one or the other wire and earth.
In case of an automated routine test, the polarity of the DC voltage shall be the same as for the normal line voltage. If a possible fault is detected, a following test with reversed polarity may be performed.
In case of manual tests for the benefit of fault analysis, measurements are performed with both polarities of the line voltage.

8.3.2 Line voltage during tests and measurements

During tests the line voltage at the network termination may be between 0 V and 100 V. During some time no voltage shall be supplied in order to test for possible improper voltages on the line due to faults, but most of the time a test voltage is supplied to one or both wires for measuring the physical and electrical behaviour of the line as seen from the exchange.
As a result of manual tests in case of faults or maintenance activities, the line voltage may be switched off for a relative long time. It is advisable that terminal equipment, which depends for some function (e.g. storage of information) on continuous feeding voltage, can cope with such feeding interruption. It is also advisable that terminal equipment will return automatically to the normal operational condition after the normal line voltage has returned back.

8.3.3 Availability during tests and measurements

During the tests and measurements the subscriber line is not available for calls.
Automatic routine tests last for such a short time that the user shall have a little chance to experience them. No routine test shall be started on a line, which is detected to be occupied for a call.
Manual tests are only performed in case of fault repair activities; often the customer is aware of such situation.

8.3.4 Voltage variations and polarity reversals

As described before, tests and measurements are accompanied by voltage variations and polarity reversals. As routine tests are mainly performed during night when the call intensity is low, users will experience it annoying when there terminal equipment would react audibly on such variations. So it is advised that the ringing signal detection circuitry of TE does not act under the before mentioned conditions, i.e. that the audible alerting circuitry will not be activated; see section 5.4 for a voluntary requirement about the ringing signal detector insensitivity.