DESCRIPTION

Prepared
Rustan Gandvik, Göran Engström

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NWSC

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DESCRIPTION OF NETWORK INTERFACES ; ANALOGUE ACCESS TO PSTN

Telia AB (publ)

Telia Carrier & Networks, Network Solutions

1 INTRODUCTION

This document has been produced by Telia in order to meet the interface publication responsibility as set forth in Article 4.2 of the R&TTE directive [1]. The document describes the network interface functionality of the Analogue single line interface used for the PSTN access service as provided by Telia.

The information in this document is intended to assist the designers of telecommunications terminal equipment capable of using the services provided at the interface. The document is based on TR 101 730 [2].

The document is applicable for analogue interfaces connected to different types of network equipment delivering the PSTN service.

Interfaces for connection of Private Branch Exchanges (PBXs) and some other types of terminal equipment (e.g. payphones) will be separately described.

The characteristics of the network is defined at the **Network Termination Point (NTP)** which is the physical point at the boundary of the telephone network intended to accept the connection of a terminal equipment.

2 TERMINOLOGY AND ABBREVIATIONS

AC	Alternating Current
DC	Direct Current
DTMF	Dual Tone Multi Frequency
NTP	Network Termination Point
PSTN	Public Switched Telephone Network
TE	Terminal Equipment

3 REFERENCES

[1] Directive 1999/5/EC of the European Parliament and of the Council of 9 March 1999 on radio equipment and telecommunications terminal equipment and the mutual recognition of their conformity.

[2] ETSI TR 101 730 ; “Publication of interface specifications under Directive 1999/5/EC ; Guidelines for describing analogue interfaces.”

[3] Swedish Standard SS 455 15 50 ; “Anslutningsdon för terminaler i telenät ; Tele-terminal connectors”

[4] ETSI EG 201 120 ; “Public Switched Telephone Network (PSTN) ; Method of rating terminal equipment so that it can be connected in series and/or in parallel to a Network Termination Point (NTP)”

[5] ETSI ES 201 187 ; “2-wire analogue voice band interfaces ; Loop Disconnect (LD) dialling specific requirements”

[6] ETSI ES 201 235-1 , -2 & -3 ; “Specification of Dual Tone Multi-Frequency (DTMF) Transmitters and Receivers”
Part 1: General Part 2: Transmitters Part 3: Receivers

[7] ETSI EN 300 659-1 ; “Public Switched Telephone Network (PSTN) ; Subscriber line protocol over the local loop for display (and related) services ; Part 1: On-hook data transmission” ***(draft)***

[8] Telia specification 8211-A 331 ; “Transfer of number information by means of DTMF on analogue subscriber lines in Telia access network”

[9] ITU-T Recommendation O.9 ; “Measuring arrangements to assess the degreee of unbalance about earth”

[10] ITU-T Recommendation E.180 ; “Technical characteristics of tones for the telephone service”

4 CONNECTION METHODS

The NTP for PSTN is usually a 4-pin Swedish type of socket according to Swedish Standard SS 455 15 50 [3] but other sockets e.g. RJ11/12 and RJ45 are also in use. Normally only the first socket is installed by the network provider. Further sockets within the user installation may be added and installed by an installation company or by the user. No termination (e.g. RC-network) for testing the line is provided.

4.1 Socket/plug of the Swedish type

4.1.1 Installation with sockets connected in cascade

The user installation, which connects TEs to the PSTN, may consist of several sockets of the Swedish type wired according to figure 1. Contacts 1 and 2 are used for connection to the PSTN and contacts 3 and 4 are used for connection of a line to the next socket. Each socket also incorporates a switch (contacts 5 and 6) to bypass the socket when not used.

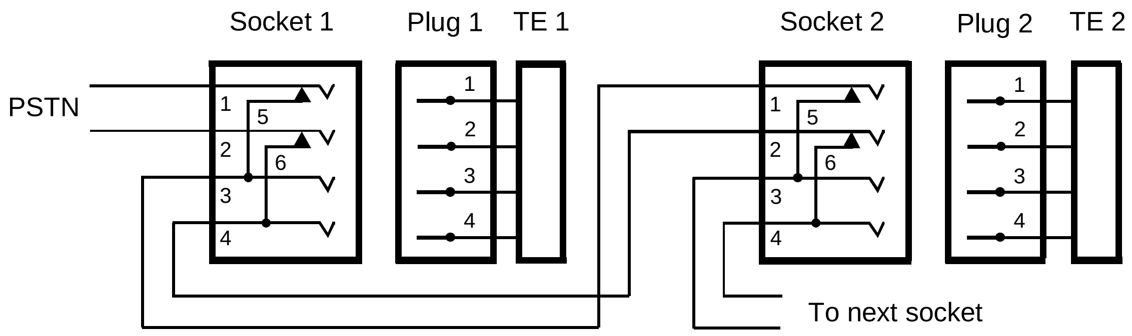

Figure 1 : Wiring of sockets (SS 455 15 50)

The contacts are arranged in such a way that TEs designed for 4-wire connection will be connected in cascade, in accordance with the two-port principle, see figure 2.

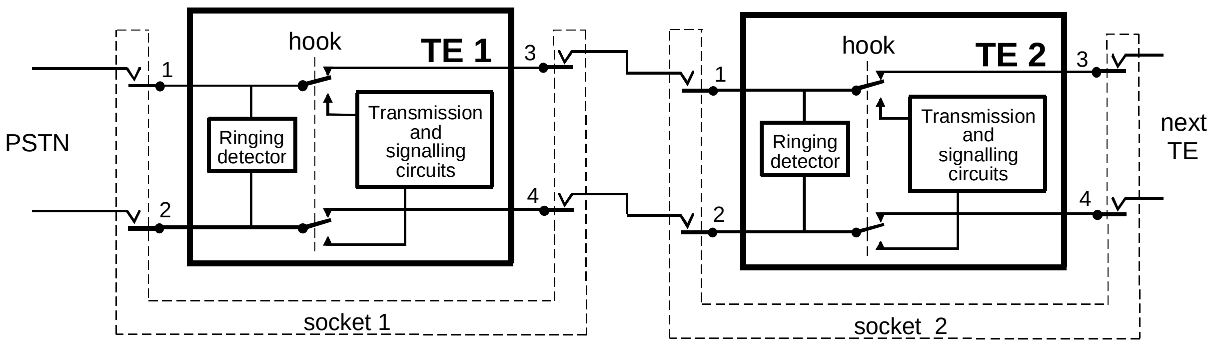

Figure 2 : 4-wire TEs connected in cascade

Characteristics of the cascade installation with TEs designed for 4-wire connection:

- If all TEs in the user installation are in quiescent state, they are all connected simultaneously in parallel to the line.
- More than one TE can not be in loop state simultaneously.
- A TE can not be placed in loop state if it is connected to a socket behind another TE in loop state.
- A TE in loop state will be disconnected if another TE, connected to a socket prior to the TE first mentioned, is placed in loop state.

Also TEs designed for 2-wire connection, may be connected to this type of installation.

4.1.2 Installation with sockets connected in parallel

The user installation may as an alternative to clause 4.1.1 consist of several sockets connected in parallel. Only contacts 1 and 2 of each socket, see figure 1, are used in this case.

TEs designed for 2-wire connection, may be connected to this type of installation.

TEs designed for 4-wire connection, may be connected to this type of installation, although the characteristics will differ from that of the cascade installation described in clause 4.1.1.

4.1.3 Wiring of TE and plug

A TE designed for 2-wire connection to the Swedish type of socket shall be connected to contacts 1 and 2 of the Swedish type of plug, see figure 3. With such a TE connected to a cascade installation (see figure 1) all sockets behind the TE are disconnected. To preclude such a disconnection, as it is usually not intended, there should be links between contacts 1 and 3 and contacts 2 and 4. These links can be placed within the plug or, if the TE is provided with a 4-wire cord, within the TE.

A TE designed for 4-wire connection (see figure 2) to the Swedish type of socket shall incorporate a switch that,

- in quiescent state, connects the socket behind the TE to the line
- in loop state, disconnects the socket behind the TE from the line.

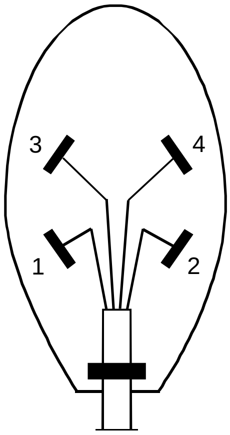

Figure 3 : Swedish type of plug (SS 455 15 50)

4.2 Socket/plug of the type RJ11/12

4.2.1 Installation with sockets connected in parallel

The user installation, which connects TEs to the PSTN, may consist of one or several sockets of the type RJ11/12 connected in parallel using contacts 3 and 4 in each socket (see figure 4).

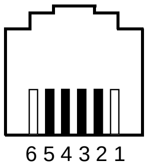

Figure 4 : RJ11/12 socket, front view

4.2.2 Installation with sockets connected in cascade

The user installation may as an alternative to clause 4.2.1 consist of several sockets connected in cascade using contacts 3 and 4 for the connection towards the PSTN and contacts 5 and 2 for the connection of a line to the next socket. In such an installation the connection to sockets behind a certain socket will be disconnected if this socket is not used. This can be precluded by inserting a dummy plug in this socket.

4.2.3 Wiring of TE and plug

A TE designed for 2-wire connection to the installation described in clause 4.2.1 shall provide a plug of the type RJ11/12 using contacts 3 and 4.

A TE designed for 4-wire connection to the installation described in clause 4.2.2 shall provide a plug of the type RJ11/12 using contacts 3 and 4 for connection towards the PSTN and the contacts 5 and 2 for the connection of a line towards the next socket.

4.3 Socket/plug of the type RJ45

4.3.1 Installation with sockets connected in parallel

The user installation, which connects TEs to the PSTN, may consist of one or several sockets of the type RJ45 connected in parallel using contacts 4 and 5 in each socket (see figure 5).

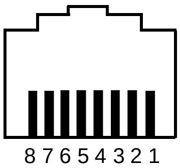

Figure 5 : RJ45 socket, front view

4.3.2 Installation with sockets connected in cascade

The user installation may as an alternative to clause 4.3.1 consist of several sockets connected in cascade using contacts 4 and 5 for the connection towards the PSTN and contacts 3 and 6 for the connection of a line to the next socket. The contacts 1, 2, 7 and 8 are not used. In such an installation the connection to sockets behind a certain socket will be disconnected if this socket is not used. This can be precluded by inserting a dummy plug in this socket.

4.3.3 Wiring of TE and plug

A TE designed for 2-wire connection to the installation described in clause 4.3.1 shall provide a plug of the type RJ45 using contacts 4 and 5.

A TE designed for 4-wire connection to the installation described in clause 4.3.2 shall provide a plug of the type RJ45 using contacts 4 and 5 for connection towards the PSTN and the contacts 3 and 6 for the connection of a line towards the next socket.

5 WIRING ARRANGEMENTS AND DRIVING CAPABILITY

Wiring arrangements is described in clause 4. The NTP is capable of supporting a combination of TEs whose total load is not greater than 100 LU as specified in ETSI EG 201 120 [4].

6 DC VOLTAGES AND FEED CONDITIONS

6.1 DC Voltage – On Hook

DC voltage across 100 kohms placed across a- and b-wire at the NTP in the PSTN :

Maximum : 55 V
Minimum : 45 V

Maximum DC voltage used for line testing: 100 V

6.2 Polarity

Idle state polarity is defined as (+) on a-wire and (–) on b-wire.
Conversation state polarity is defined as (-) on a-wire and (+) on b-wire.

Which is a-wire and b-wire at the NTP, may vary from time to time depending on operator´s maintenance activities in the access network.

Polarity reversals are used as Answer Signal (see clause 10) and as Clearing Signal from the network (see clause13). Polarity reversals are also used as alerting signals in conjunction with display services (see clause 15).

6.3 Line current

6.3.1 DC current supplied using resistive feeding

The majority of NTPs are supplied by current from feeding bridges consisting of nominal 48-50 V in series with nominal 2x800 ohms. The line length between the feeding bridge and the NTP is normally limited to a loop resistance of 1200 ohms. The maximum and minimum currents at the NTP will depend on the DC resistance of the user´s TE.

Maximum DC current at 0 ohms loop resistance and when a 0 ohms load is placed across the a- and b-wires at the NTP : 36 mA

Minimum DC current at 1200 ohms loop resistance and when a 300 ohms load is placed across the a- and b-wires at the NTP : 14 mA

6.3.2 DC current supplied using constant current

A minor part of the NTPs are supplied by constant current feed supplies. At present these are used for short local loop applications (loop resistance 0-300 ohms) e.g. ISDN or IP terminal adapters and pair gain systems. These systems are specified to provide a current of 25-29 mA when the NTP is loaded with 200-900 ohms DC resistance.

6.4 Park condition

Certain types of line interfaces may enter a park condition (power-down mode) resulting in a constant current feed of 14-15 mA.

This will apply at an NTP where the connected TE

- is still off-hook and 60s has elapsed since the clearing signal (see clause 13) from the network was sent.
- has seized the line but no dialling has started and 90 s has elapsed.

7 SEIZURE

7.1 Conditions at the NTP not recognized as a seizure

A resistance of >12 kohms placed across the a- and b-wires at the NTP, will not be recognised as a seizure condition. It should be noted that in general the resistance of the user´s installation in quiescent state is substantial lower than the resistance of a single TE as the installation consist of several TEs connected in parallel.

7.2 Conditions at the NTP to facilitate line testing

To facilitate line testing, the DC resistance between the a- and b-wires of a TE should be substantially higher than the non-seizure value of clause 7.1.

The minimum DC resistance that may be placed across the a- and b-wires at the NTP without disturbing the line testing equipment is 1 Mohms.

The minimum DC resistance that may be placed between a-wire and earth or between b-wire and earth at the NTP without disturbing the line testing equipment is 10 Mohms.

These parameters will be affected by the number of TEs connected in parallel.

7.3 Conditions at the NTP which will be interpreted as a seize signal

A resistance of <1,3 kohms placed across the a-wire and b-wire at the NTP, will be recognised as a seizure condition.

7.4 Time required for seizure signal to be recognised

A seizure condition that is applied across the a-wire and b-wire during < 80 ms will not be interpreted as a seizure condition.

A seizure condition that is applied across the a-wire and b-wire during >120 ms will be interpreted as a seizure condition.

8 DIALLING

8.1 Type of dialling accepted

All analogue single line PSTN interfaces support DTMF dialling. Loop disconnect dialling is so far supported by the majority of interfaces. However, it is not recommended to use loop disconnect dialling in new TE as

- an increasing number of PSTN interfaces will not support this dialling
- the loop disconnect dialling as used by Telia deviates from ES 201 187 [5]
- supplementary services and other DTMF based services can not be used.

8.2 Reception of first digit

The network is ready to receive the first digit during the time period when dial tone is sent (see clauses 18.1 and 18.2). For some type of interfaces, dial tone is presented 150-250 ms after the line seizure. Other types of interfaces perform a line test before sending dial tone. In that case, dial tone sending may be delayed up to 1 s.

8.3 Number and timing of call attempts

The network will accept a minimum time interval of 0,5 s between the release by the TE of one (unsuccessful) call attempt and the seizure for the next attempt.

NOTE : In most practical applications it is however sensible to use a considerably greater value, so as to provide an appropriate compromise between the rate of redialing and the likelihood of the repeat call attempt being successful, taking into account the typical holding times for different types of calls.

It is recommended to limit the number of call attempts in a redial sequence to reduce the disturbances that will occur, if a wrong number is dialled repeatedly.

8.4 DTMF dialling

At the NTP the network will accept DTMF transmitters meeting the characteristics of ES 201 235-2 [6].

8.5 Loop disconnect dialling

Not recommended. See clause 8.1.

9 RINGING SIGNALS

The *cadence* of a ringing signal sent by the PSTN is the pattern of sound/silence which gives it a characteristic rhythm. E.g. for a signal with a cadence of 1,0 s on and 5,0 s off, the cadence is denoted as **1,0** – 5,0 s.

9.1 Ordinary ringing signal

Ordinary ringing signal is sent to the called party to indicate an incoming call.

Cadence : **1,0** – 5,0 s tolerance : ± 10 %

However, the first ringing pulse may have a duration of 1,0-1,6 s

Duration : As long as the call has not been answered by the called party or released by the calling party, however not more than 180 s.

Frequency : 25 Hz tolerance : ± 2 Hz

Maximum AC voltage : 90 Vrms

Minimum AC voltage across a load of 4 kohms : 40 Vrms

The waveform is sinusoidal with harmonic distortion < 10 %
at resistive loads > 4 kohms.

The ringing voltage is symmetrically superimposed on the idle state DC feeding voltage.

When the TE has answered the call at the NTP, the ringing signal may be applied for a further period of 200 ms.

9.2 CCSS ringing signal (recall)

The CCSS (Call Completion Supplementary Service) ringing signal is sent to a user to indicate that a called party, that previously were busy, now is available. The network has been monitoring the called party since the failed call attempt and alerts the original calling party.

Cadence: **0,3** – 0,4 s tolerance: ± 10 %

Duration : As long as the call has not been answered by the receiver of the CCSS ringing signal, however not more than 20 s.

All other characteristics: As for ordinary ringing signal, see clause 9.1

10 ANSWER SIGNAL

The PSTN indicates call answer by reversing the voltage polarity at the NTP of both the calling and the called party (from idle to communication state polarity). The polarity reversal at the called party NTP normally takes place 100 ms after the answer.

11 CHARGING INFORMATION

The charging information service in the PSTN based on sending of 12 kHz pulses has been phased out and is available only for existing users of the service.

12 REGISTER RECALL

Terminal equipment generates a Register Recall signal (R-signal) by breaking the normal DC loop for a specific period (timed break recall). The PSTN will accept register recall signals meeting the following description:

Minimum duration of the break period :	50 ms
Maximum duration of the break period:	130 ms
Maximum residual current during break period:	1 mA

13 CLEARING SIGNAL FROM THE NETWORK

When in communication state, the PSTN has detected a high-ohmic loop (TE on-hook) at the NTP of the ***calling party***, the PSTN will disconnect the call and send a release signal i.e. reverse the voltage polarity from communication to idle state polarity at both the calling and the called party NTP within 3 s.

When in communication state, the PSTN has detected a high-ohmic loop (TE on-hook) at the NTP of the ***called party***, the PSTN will disconnect the call and send a release signal i.e. reverse the voltage polarity from communication to idle state polarity at both the calling and the called party NTP after 90 s.

During the supervision time (the 90 s), the called party may return to communication state by connecting a low-ohmic loop (TE off-hook) to the NTP. In case of a subsequent high-ohmic loop the 90 s timer will restart. If, during the supervision time the PSTN detects a high-ohmic loop (TE on-hook) at the NTP of the calling party, the PSTN will disconnect the call and send a release signal i.e. reverse the voltage polarity from communication to idle state polarity at both the calling and the called party NTP within 3 s.

NOTE: The supervision time 90 s, that makes it possible for the called party to change to another TE without releasing the call, will be reduced to 45 s in the future.

14 SIGNALLING FOR SUPPLEMENTARY SERVICES

The supplementary services mentioned below are available in the PSTN. Some of them are generally available and free, some have to be requested from Telia at certain price. For activation and deactivation of the services, DTMF characters including * and # shall be used. For making switching orders, the Register Recall (R) function shall be used (see clause 12).

14.1 Services with standardized service codes

14.1.1 Call Forwarding Unconditional (CFU)

CFU (Swedish term: *Vidarekoppling direkt*) enables a user to have all incoming calls, which are addressed to his number, forwarded to another number.

CFU to any number :

Activation :	* 21 * number #		
Deactivation :	# 21 #		
Status check :	* # 21 #	or	* # 21 * number #

CFU to a fixed number. This fixed number has to be set by Telia :

Activation :	* 22 #
Deactivation :	# 22 #
Status check :	* # 22 #

All actions are confirmed by verbal recorded announcements from the network.

14.1.2 Call Forwarding on No Reply (CFNR)

CFNR (Swedish term: *Vidarekoppling vid ej svar*) enables a user to have all incoming calls, which meet with no reply and are addressed to his number, forwarded to another number.

CFNR to any number :

Activation :	* 61 * number #	(forwarding after 28 s)
or :	* 61 * number * ss #	where ss (5-60 s) is the time until forwarding
Deactivation :	# 61 #	
Status check :	* # 61 # or * # 61 * number #	

CFNR to a fixed number. This fixed number has to be set by Telia :

Activation :	* 62 #	(forwarding after 28 s)
or :	* 62 * ss #	where ss (5-60 s) is the time until forwarding
Deactivation :	# 62 #	
Status check :	* # 62 #	

All actions are confirmed by verbal recorded announcements from the network.

14.1.3 Call Forwarding on Busy (CFB)

CFB (Swedish term: *Vidarekoppling vid upptaget*) enables a user to have all incoming calls, which meet with busy and are addressed to his number, forwarded to another number. This supplementary service has to be requested from Telia.

CFB to any number :

Activation :	* 67 * number #
Deactivation :	# 67 #
Status check :	* # 67 # or * # 67 * number #

CFB to a fixed number. This fixed number has to be set by Telia.

Activation :	* 68 #
Deactivation :	# 68 #
Status check :	* # 68 #

All actions are confirmed by verbal recorded announcements from the network.

14.1.4 Abbreviated dialling (ADI)

ADI (Swedish term: *Kortnummer)* enables a user to make a call by sending a short code instead of a full number. This supplementary service has to be requested from Telia.

Registration of a short code:	* 51 * short code * full number #
Removal of a short code:	# 51 * short code #
Status check:	* # 51 * short code # or
	* # 51 * short code * full number #

All actions are confirmed by verbal recorded announcements from the network.

To use the short code : short code #

14.1.5 Fixed destination call (FDC)

FDC (Swedish term: *Direkt uppringning)* enables a user to set up a call to a predetermined number, by lifting the handset only. This supplementary service has to be requested from Telia. Three variants exist: a) Immediate set up, b) set up with 5s delay and c) set up with 12s delay. The variants b) and c) may be managed by the user as follows :

Registration :	* 53 * the predetermined number #
Removal :	# 53 #
Status check :	* # 53 # or * # 53 * the predetermined number #

All actions are confirmed by verbal recorded announcements from the network.

14.1.6 Completion of Calls to Busy Subscriber (CCBS)

CCBS (Swedish term: *Återuppringning vid upptaget)* enables a calling user, encountering a busy destination, to have the call completed when the busy destination becomes idle, without having to make a new call attempt.

Activation :	press 5 when encountering a busy tone	
Deactivation :	# 37 #	Deactivates all CCBS
or :	# 37 * number #	Deactivates CCBS to certain number
Status check :	* # 37 # or * # 37 * number #	

All actions are confirmed by verbal recorded announcements from the network.

The PSTN alerts the original calling user with a CCSS ringing signal (see clause 9.2) when the busy destination becomes idle, if within 45 minutes. When the original calling user answers , the former busy destination will be alerted by an ordinary ringing signal.

14.1.7 Call Waiting (CAW)

CAW (Swedish term: *Samtal väntar*) enables a busy user to be notified of a new incoming call that is in a waiting position. The user then has the choice of accepting, rejecting or ignoring the waiting call, making use of switching orders based on R (register recall according to clause 12).

Activation :	* 43 #
Deactivation :	# 43 #
Status check :	* # 43 #

All actions are confirmed by verbal recorded announcements from the network.

Switching Orders that are available when the busy user is notified of a new incoming call by the alerting signal according to 18.6 :

To reject the new call without answering it : R0 [1)]

To release the old call and take the new call : R1

To place the old (current) call on hold and take the new call : R2

To switch between the old and the new call : R2

To connect to both the old and the new call : R3

1) The new calling party will receive Ringing Tone (se clause 18.7) which will be replaced by Busy Tone (se clause 18.3) after 24s if the busy user ignores the alerting signal. If the busy user rejects the new call (by pressing R0), the calling party will receive Busy Tone at that instant.

14.1.8 Last Number repetition (LNR)

LNR (Swedish term: *Repetition av senast slaget nummer)* enables a user to repeat the last number dialled by dialling a short code. This supplementary service has to be requested from Telia.

Code to dial the last number : * * 0

14.1.9 Enquiry service (ENQ)

ENQ (Swedish term: *Förfrågan/pendling*) enables a user to interrupt communications on an existing call, make a new call and then subsequently, switch between the old and new call.

Procedure: Two parties are engaged in a call. One of the parties (the active party) places the other on hold by pressing R. The active party receives dial tone and makes a call to a new party. After the new party has replied, the active party may return to the old party by pressing R1 and switch between the old and new party by pressing R2. If the new party does not reply, the active party may stop the call attempt and return to the party on hold by pressing R.

14.1.10 Conference call, 3-party (3PTY)

3PTY (Swedish term: *Trepart*) enables a user to establish a 3-party conversation. The service can be invoked during the call waiting service (see clause 14.1.7) but can also be established by a separate procedure.

Procedure is as for Enquiry service (see clause 14.1.9) but with the following addition: When the new party has replied, the active party may press R3 to connect all three parties to the call.

14.1.11 Alarm call, casual (ALS)

ALS (Swedish term: *Väckning/Påminnelse)* enables a user to place an alarm call to be made to his line within the next 24 hours, at a time specified (as hhmm =hours and minutes) in advance by him.

Registration :	* 55 * hhmm #		
Removal of all ALS:	# 55 #		
Removal of certain ALS :	# 55 * hhmm #		
Status check :	* # 55 #	or	* # 55 * hhmm #

All actions are confirmed by verbal recorded announcements from the network.

14.1.12 Calling Line Identification Restriction (CLIR)

CLIR (Swedish term: *Skydd mot nummerpresentation*) enables a calling party to prevent presentation, on a call by call basis, of his number to the called party.

Activation: # 31 # is dialled immediately before the called party number.

14.1.13 Outgoing Call Barring (OCB)

OCB (Swedish term: *Räckviddsbegränsare*) enables a user to prevent all or certain types of outgoing calls. This supplementary service has to be requested from Telia who also provides the Personal Identification Number (PIN).

OCB with fixed barring areas :

Activation : * 33 * PIN #

Deactivation : # 33 * PIN #

Status check : * # 33 #

OCB with user selection of barring areas :

Activation : * 34 * PIN * parameter defining the barring area #

Deactivation : # 34 * PIN #

Status check : * # 34 #

All actions are confirmed by verbal recorded announcements from the network.

14.2 Services with national service codes

14.2.1 Telia TeleSvar

Telia TeleSvar is a network based answering machine facility that can leave, take and play back messages. TeleSvar has a capacity for up to 100 messages and each message may be up to five minutes long. This supplementary service has to be requested from Telia.

a) Control of TeleSvar from the home number (for which the service has been ordered):

Connect to TeleSvar : * 133 #

This gives access to the main menu where the user by using 0→9 and # may select to play messages, make pauses, repeat, step between messages, delete messages, undo deletions, record/listen to own greeting messages, activate/deactivate time recordings and set/change the personal 4 digit code.

Set the number of ringing signals before the TeleSvar answers : * 132 * N # where N=1→9

Set the number of ringing signals before the TeleSvar answers to default value, N=4 : * 132 * #

Disconnect TeleSvar : # 132 #

Set TeleSvar for immediate answer : *131 #

Disconnect TeleSvar : # 131 #

b) Control of TeleSvar from another number in Telia PSTN

Connect to TeleSvar : * 137 # home number
interrupt the greeting by : 0 personal code #

This gives access to the main menu as in a) above plus possibility to change the number of ringing signals before the answer and connect/disconnect TeleSvar.

c) Control of TeleSvar from an arbitrary number

Connect to TeleSvar : home number
interrupt the greeting by : 0 personal code #

This gives access to the main menu as in b) above.

15 SIGNALLING FOR PSTN DISPLAY SERVICES

CLIP (Calling Line Identification Presentation) may be provided at the user´s request at all types of NTPs in the PSTN. The method used for the transfer is the DTMF protocol according to annex B of EN 300 659-1 [7]. The procedure includes polarity reversals.

Description of the sending procedure:

1. The voltage polarity at the NTP of the ***called party*** is reversed from idle to communication state polarity (see 6.2) to alert the terminal.

2. 180-1000 ms after the reversal, the CLIP sending starts by means of DTMF.

3. The ordinary DTMF sequence sent is D-S_1-S_2...S_n-C , where
   - D and C are start and stop codes, respectively
   - S_1-S_2...S_n is the calling number transferred as
     - 0 + trunk code + subscriber number

     or in case of an international call :
     - 00 + country code + trunk code + subscriber number

   In case of a diverted call, it is not the calling number but the redirecting number that is transferred.

4. There are plans for a general change to use the sequence A-S_1-S_2...S_n-C for indication of ordinary calls (incl. calling number) and the sequence D-S_1-S_2...S_n-C for indication of diverted calls (incl. redirecting number). Some types of PSTN interfaces already have this feature.

5. There are also plans to transfer the sequence A-S_1-S_2...S_n-D-S_1-S_2...S_n-C for diverted calls, indicating both calling number and redirecting number..

6. If no number is delivered, the sequence B-t_1-t_2-C is transferred, where
   - t_1-t_2 = 0-0 indicates that no number is available
   - t_1-t_2 = 1-0 indicates that the number is restricted from presentation

7. Immediately after the DTMF sequence, the voltage polarity is reversed from communication state polarity to idle polarity (see clause 6.2).

8. Ringing signal is applied within 1 s from sending of the stop code C.

9. The DTMF codes A, B, C and D are normally not of common knowledge to the user and should not be presented.

10. The DTMF codes, frequencies, tone and pause durations are according to ES 201 235-1 and –2 [6].

11. Further information may be found in the Telia specification 8211-A 331 [8].

12. The power level of each of the two signalling frequencies, producing each DTMF code, across a reference impedance Z_R (see clause 16.1) placed at the NTP will fall between –7 dBm and –27 dBm. Normally a TE with a DTMF receiver for CLIP will have a much higher impedance than Z_R which may result in levels at NTP up to -2 dBm.

16 TRANSMISSION

16.1 Reference impedance and level definition

Transmission characteristics at the NTP are, if not otherwise mentioned, defined when the line is terminated at the NTP with the European reference impedance Z_R = 270 ohms + 750 ohms // 150 nF.

The signal levels are defined in terms of dBm across Z_R.

A level of W dBm corresponds to a voltage of $U=\sqrt{0{,}001\times\left|Z_{1020}\right|\times 10^{W/10}}$ volts,

where $\left|Z_{1020}\right|$ is the modulus of Z_R at 1020 Hz (i.e. 842 ohms).
Consequently a power level of 0 dBm corresponds to a voltage of 918 mV.

16.2 Relative levels

Relative input level (Li) at 1020 Hz at the NTP :	0 dBr to +12 dBr
Relative output level (Lo) at 1020 Hz at the NTP :	-5 dBr to -17 dBr

16.3 Frequency band

The transmission channel has the capability to transfer the frequency range 300 Hz – 3400 Hz.

16.4 Attenuation distortion

The variation with frequency of the attenuation between the NTP and a digital interface of the PSTN will fall within the mask shown in figure 6. The distortion is defined relative to the attenuation at 1020 Hz.

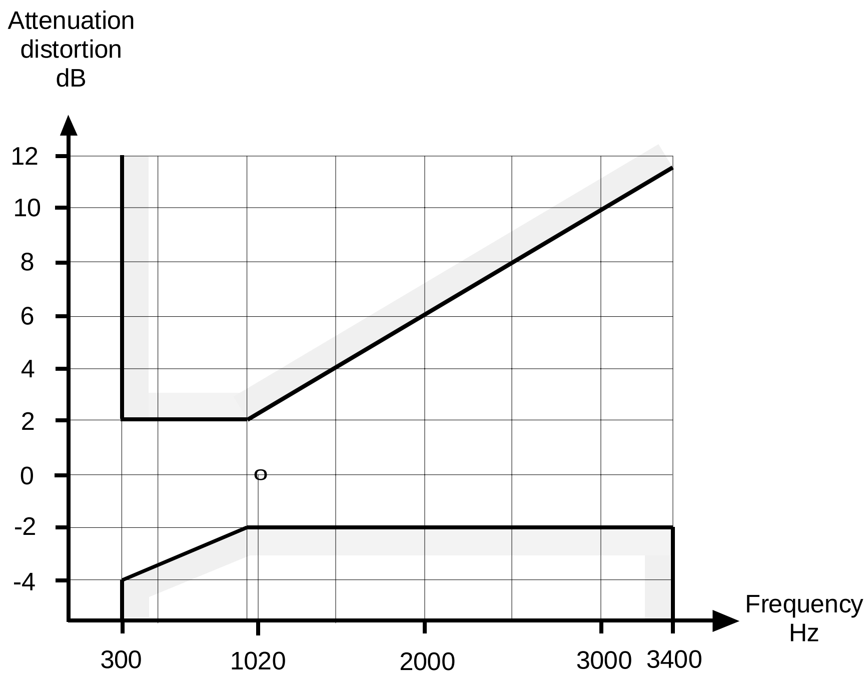

Figure 6 : Attenuation distortion

16.5 Send and receive loudness rating

The recommended nominal send loudness rating (SLR) and receive loudness rating (RLR) for voice terminals connected to the NTP are
SLR = +3 dB and RLR= -8 dB

16.6 Input impedance

The input impedance of the PSTN as seen at the NTP will depend on frequency, the line characteristics and the input impedance of the exchange termination.

The nominal input impedances of the PSTN local exchange terminations are

- 600 ohms
- 900 ohms // 60 nF
- 275 ohms + 850 ohms // 150 nF, or
- 270 ohms + 750 ohms //150 nF

The loop resistance of a copper pair between the NTP and the local exchange is less than 1200 ohms. A copper pair consists of sections of different cables having nominal characteristics according to the table below.

conductor diameter	resistance	capacitance
0,4 mm	280 ohms / km	45 nF/km
0,5 mm	178 ohms / km	45 nF/km
0,6 mm	123 ohms / km	45 nF/km
0,7 mm	91 ohms / km	45 nF/km

The input impedance of the PSTN at the NTP can be calculated based on the values above. It is estimated that the lowest value of return loss, relative to the reference impedance Z_R, within the frequency range 300-3400 Hz will be 8 dB.

16.7 Longitudinal conversion loss

Longitudinal Conversion Loss (LCL) as defined in ITU-T Recommendation O.9 [9] at the NTP is

> 40 dB in the frequency range 300-600 Hz, and

> 46 dB in the frequency range 600-3400 Hz.

16.8 Noise

The psophometrically weighted output noise at the NTP when terminated by Z_R is less than –60 dBmp.

The level of any single frequency component in the 300-3400 frequency range, corrected by the psophometric weighting factor, is less than –70 dBm.

17 ACCEPTABLE SIGNAL LEVELS AT THE NTP

In order to ensure that signals will be transmitted correctly and that the signals do not interfere with other circuits, levels exceeding those stated below should not be transmitted by the TE into the PSTN at the NTP. All levels are defined across Z_R (see clause 16.1).

17.1 Mean sending levels

The highest acceptable mean sending power level in the frequency range 200-3800 Hz over a one-minute period is –9,0 dBm. This does **not** apply to DTMF signals.

17.2 Instantaneous sending voltage

The highest acceptable peak to peak voltage is 5,0 V.

17.3 Sending levels in a 10 Hz bandwidth

The sending power level in a 10 Hz bandwidth centred at any point in the frequency range 30 Hz to 4300 Hz (and wholly contained within that range) should not exceed the limits given in figure 7. This does **not** apply to DTMF signals.

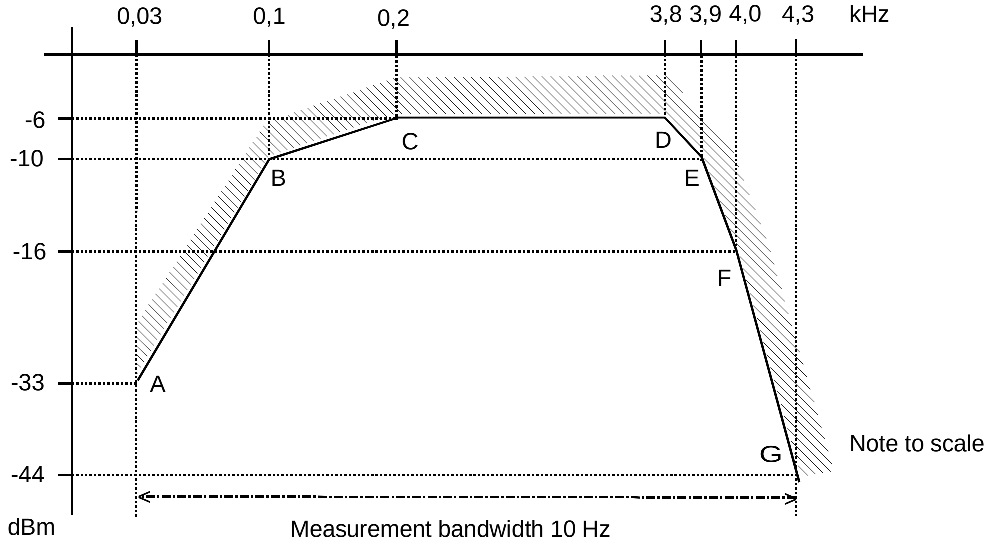

Figure 7

17.4 Sending levels above 4,3 kHz

The sending power level in a bandwidth defined in figure 8, centred at any point in the frequency range 4,3-200 kHz (and wholly contained within that range) should not exceed the limits given in figure 8. This does **not** apply to DTMF signals.

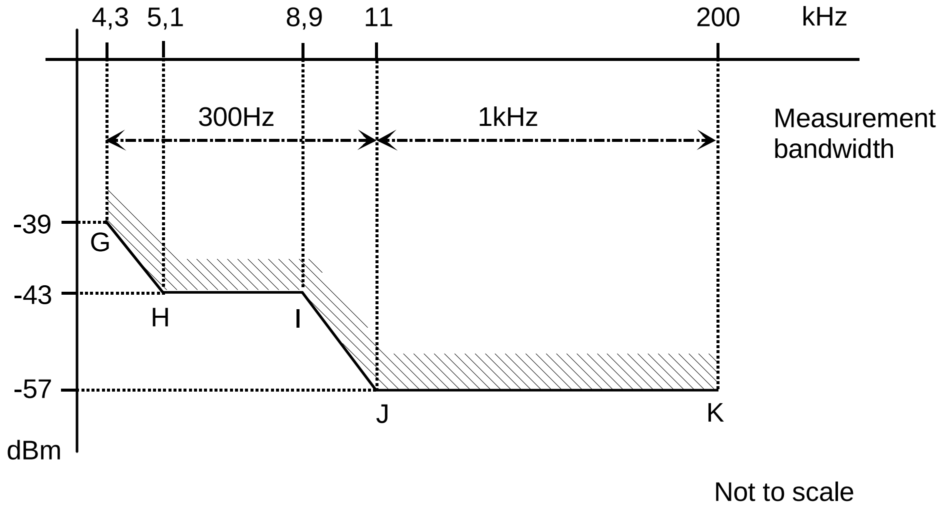

Figure 8

18 SUPERVISORY TONES

In this clause are described the audible information tones that are sent by the PSTN to inform the user about the state of a telephone call or a supplementary service. In each subclause is specified whether the PSTN sends the tone to the calling party or to the called party. Each information tone is a sound composed of one or several frequencies.

The *cadence* is the pattern of sound/silence in a tone which gives it a characteristic rhythm. E.g. for a tone with a cadence of 1,0 s on and 5,0 s off, the cadence is denoted as **1,0** – 5,0 s.

The *level* of the tones is defined in terms of dBm across the reference impedance Z_R as defined in clause 16.1.

18.1 Dial tone

Dial tone (Swedish term: *Kopplingston)* is sent to the calling party to indicate that the network is ready to receive call information and inviting the user to start sending call or service related information.

Cadence:	**continuous**	
Duration:	15 s	If dialling not is started within 15 s, the dial tone is replaced by congestion tone or silence.
Frequency:	425 Hz ± 15 Hz	(sinusoidal)
Level:	-8 dBm to -22 dBm	

18.2 Special dial tone

The special dial tone (Swedish term: *Speciell kopplingston)* is sent to the calling party to indicate that the network is ready to receive call information and inviting the user to start sending call or service related information, at the same time reminding the user that special conditions apply to the terminal from which the call is being made.

Cadence:	**0,32** – 0,02 s ±10 %	
Duration:	15 s	If dialling not is started within 15 s, the special dial tone is replaced by congestion tone or silence.
Frequency:	425 Hz ± 15 Hz	(sinusoidal)
Level:	-8 dBm to -22 dBm	

18.3 Busy tone

Busy tone (Swedish term: *Upptagetton)* is sent to the calling party to indicate that a connection has been made but that the called party is busy.

Cadence: **0,25** – 0,25 s ±10 %
Duration: 30 s or 15 s
Frequency: 425 Hz ± 15 Hz (sinusoidal)
Level: -13 dBm to -27 dBm

NOTE: The difference in cadence between the Telia busy tone and congestion tone is contradictory to what is recommended in the ITU-T Recommendation E.180 [10].

18.4 Congestion tone

Congestion tone (Swedish term: *Spärrton)* is sent to the calling party to indicate that some part of the network required for setting up of the requested call or for the use of a specific service is temporarily engaged.

Cadence: **0,25** – 0,75 s ±10 %
Duration: 15 s
Frequency: 425 Hz ± 15 Hz (sinusoidal)
Level: -13 dBm to -27 dBm

NOTE: The difference in cadence between the Telia congestion tone and busy tone is contradictory to what is recommended in the ITU-T Recommendation E.180 [10].

18.5 Special information tone

Special information tone (Swedish term: *Hänvisningston)* is sent to the calling party to indicate that a connection cannot be made for some reason other than "subscriber busy" or "congestion".

Cadence: 3 × (**0,332** – 0,024) – 2,0 s
Duration: 5 sequences
Frequencies: 950, 1400 and 1800 Hz ± 50 Hz (sinusoidal)
Level: -25 dBm to -43 dBm

The tone is also used as a ”Number unobtainable tone” and is in that case followed by a verbal recorded announcement :

Cadence: 3 × (**0,332** – 0,024) s – 1,0 s – verbal announcement (4,0 s)
Duration: 2 sequences followed by 15 s of busy tone or silence
Frequencies: 950, 1400 and 1800 Hz ± 50 Hz (sinusoidal)
Level: -25 dBm to -43 dBm

18.6 Call waiting tone

Call waiting tone (Swedish term: *Samtal-väntar-ton)* is sent to a user during a call to indicate that a new call is arriving. The indication is presented when the Call Waiting supplementary service is active and a new call invokes the service. The network sends the call waiting tone to the party that the new call addresses.

Cadence:	**0,2** – 0,5 – **0,2** s ±10 %
Duration:	Only one sequence
Frequency:	425 Hz ± 15 Hz (sinusoidal)
Level:	-13 dBm to -27 dBm

18.7 Ringing tone

Ringing tone (Swedish term: *Rington)* is sent to the calling party to indicate that a connection has been made and that an alerting signal is being applied to the called terminal or service. The ringing tone is not intended to coincide with the ringing signal that is sent to the called terminal.

Cadence:	**1,0** – 5,0 s ±10 %
Duration:	As long as the call has not been answered, however a maximum of 180 s
Frequency:	425 Hz ± 15 Hz (sinusoidal)
Level:	-13 dBm to -27 dBm

18.8 Warning tone – Conference call

Warning tone – Conference call (Swedish term: *Konferenston)* is sent to a user during a call to confirm that a conferee has joined the conversation within a conference call. The warning tone is sent by the local exchange where the conference is established and to all parties of the conference call.

Cadence:	**0,33** – 15 s
Duration:	As long as the conference call is in progress
Frequency:	1400 Hz ± 50 Hz (sinusoidal)
Level:	-33 dBm to –53 dBm

18.9 Warning tone – Operator intervening

Warning tone – Operator Intervening (Swedish term: *Varningston vid telefonistpå-koppling)* is sent to a user during a call to indicate that the privacy of the conversation can no longer be assured because of intervention of an operator. The tone is sent by the local exchange where the intervention is made and to both parties of the call.

Cadence:	**0,1** – 1,5 s ± 10 %
Duration:	As long as the intervening is in progress
Frequency:	1400 Hz ± 21 Hz (sinusoidal)
Level:	-23 dBm to -43 dBm