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TELECOMMUNICATION
STANDARDIZATION SECTOR
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(10/2014)

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

Digital terminal equipments – Operations, administration and maintenance features of transmission equipment

Signal processing network equipment

Recommendation ITU-T G.776.4

ITU-T

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Recommendation ITU-T G.776.4

Signal processing network equipment

Summary

Recommendation ITU-T G.776.4 provides information and requirements for signal processing network equipment (SPNE). It can be applied to traditional networks and next generation networks (NGNs). SPNE such as network echo cancellers (ECs), voice gateways and digital circuit multiplication equipment (DCME) may include signal processing functions (SPFs) (echo cancellation, noise reduction (NR), automatic level control (ALC), circuit multiplication, etc.).

Information in this Recommendation is based on actual and planned equipment installed (or to be installed) in telecommunication networks. This Recommendation describes components, functionality, delay requirements, alarm handling and algorithms used by these devices. Transport, signalling, control, OAM and testing interfaces are also described.

History

Edition	Recommendation	Approval	Study Group	Unique ID*
1.0	ITU-T G.776.4	2014-10-14	16	11.1002/1000/12233

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FOREWORD

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

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

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

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

NOTE

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

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

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1 Scope

Signal processing network equipment (SPNE) deployed in the network (located in a transmission path from network termination unit to network termination unit) provides functions for voice enhancement and the transport of voice and voiceband services. This Recommendation describes these functions and defines performance requirements necessary for successful operation in the network that are not covered in individual SPNE Recommendations. Typical applications of SPNE are also given in this Recommendation.

2 References

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

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3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following terms defined elsewhere:

None.

3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1 acoustic echo: Acoustic echo is the reflected signal resulting from the acoustic path between the earphone/loudspeaker and microphone of a terminal, hand-held or hands-free mobile station (MS).

3.2.2 acoustic echo control: The functions performed by an acoustic echo controller (AEC).

3.2.3 cancelled end: The side of an echo canceller (EC) that contains the echo path on which this EC is intended to operate. This includes all transmission facilities and equipment (including the hybrid and terminating telephone set) which is included in the echo path.

3.2.4 signal processing network device (SPND): A physical entity (e.g., chipset) that contains one or more signal processing network functions (SPNFs).

3.2.5 signal processing network element/equipment (SPNE): A stand-alone physical entity that supports one or more SPNFs. It may also be controlled internally and/or externally and managed through a network management system (NMS) that contains managed objects, a management communications function and a management application function. SPNEs include e.g., ECs, voice enhancement devices and CME.

NOTE 1 – Examples of managed objects are: receive/transmit port, power supply, plug-in cards, SPNFs. The communications function provides facilities for the transport of telecommunications management network (TMN) messages to and from the management application function, as well as facilities for the transit of messages. The message communications function does not originate or terminate messages. A management application function is the origin and termination for all TMN messages.

NOTE 2 – A SPNE in this context is a combination of hardware and software that performs SPNFs.

3.2.6 signal processing network function (SPNF): A function within a physical entity (e.g., SPNE, SPND) that performs signal processing to provide support or services to the transport network and/or to the users. Examples include electric or acoustic echo control, NR, automatic level control (ALC), digital speech interpolation (DSI), low-rate encoding, and transcoding.

3.2.7 terminal acoustic echo controller: Terminal acoustic echo controllers are voice-operated devices installed in audio terminals on the customer premises, used for the purpose of eliminating acoustic echoes and protecting the communication from howling due to acoustic feedback from loudspeaker to microphone.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

ADPCM	Adaptive Differential Pulse Code Modulation
AEC	Acoustic Echo Controller
ALC	Automatic Level Control
BTS	Base Transceiver Station
CAS	Channel Associated Signalling
CDMA	Code Division Multiple Access

CELP	Code Excited Linear Prediction
CME	Circuit Multiplication Equipment
DCME	Digital Circuit Multiplication Equipment
DSI	Digital Speech Interpolation
DTMF	Dual-Tone Multi-Frequency
EC	Echo Canceller
EMC	Electromagnetic Compatibility
GSM	Global System for Mobile Communications
GSTN	Goods and Services Tax Network
IAD	Integrated Access Device
IP	Internet Protocol
IP-CME	Internet Protocol Circuit Multiplication Equipment
ISC	International Switching Centre
ISDN	Integrated Services Digital Network
IWF	Interworking Function
LRE	Low Bitrate Encoding
MS	Mobile Station
NLP	Non-Linear Processor
NMS	Network Management System
NR	Noise Reduction
OAM	Operation and Maintenance
PBX	Private Branch exchange
PCM	Pulse Code Modulation
PIU	Packet Interface Unit
PLC	Packet Loss Concealment
PSTN	Public Switched Telephone Network
SNMP	Simple Network Management Protocol
SPF	Signal Processing Function
SPND	Signal Processing Network Device
SPNE	Signal Processing Network Equipment
SPNF	Signal Processing Network Function
TDM	Time Division Multiplex
TDMA	Time Division Multiple Access
TFO	Tandem Free Operation
VBD-LRE	Voiceband Data-Low Rate Encoding/decoding
VED	Voice Enhancement Device
VoIP	Voice over IP

5 Conventions

None.

6 Signal processing functions

6.1 Network echo cancellation

Network echo cancellation is the process of removing cancelled-end echoes by subtracting an estimation of the cancelled-end echo. Network echo cancellers (ECs) according to [ITU-T G.168] assume that the cancelled-end is linear.

6.2 Network acoustic echo control

Network acoustic echo controllers (AECs) are devices placed in the 4-wire portion of a circuit and used for reducing cancelled-end acoustic echo. Network AECs do not necessarily assume that the cancelled-end is linear.

6.3 Comfort noise generation

Comfort noise is an artificial background noise to fill the silence in a transmission path resulting from speech gaps caused by e.g., voice activity detection or the non-linear processor (NLP) of an EC. The aim of the comfort noise generation (CNG) is to create a noise that matches the actual background noise.

If the comfort noise is well matched to that of the transmitted background acoustic noise during speech periods, the gaps between speech periods can be filled in such a way that the receiving party does not notice the switching during the conversation. Since the noise constantly changes, the comfort noise generator should be updated regularly to track the input background noise.

NOTE – The specific requirements of CNG are under study.

6.4 Noise reduction

Noise reduction (NR) is the process of reducing noise components of a signal. According to [ITU-T G.160], noise is defined as a slowly varying stochastic process appearing additive to the desired speech signal. Specifically, the variations in the characteristics of the noise process are such that it can be considered approximately stationary over much longer time intervals than a typical speech signal.

6.5 Automatic level control

Automatic level control (ALC) is a signal processing function that automatically adjusts the level of a signal towards a predetermined value.

NOTE – This function may modify the frequency response or spectral content of the signal in such a way as to affect the overall level of the signal.

6.6 Speech coding

Speech coding is a signal processing function (SPF) that compresses and decompresses speech signal streams. The purpose of this function is to reduce the size of speech signal samples and/or frames in order to reduce the bitrate. Examples are ITU-T G.711.0, ITU-T G.718, ITU-T G.726, ITU-T G.727, ITU-T G.728, ITU-T G.729, ITU-T G.722.2 and, for mobile networks, global system for mobile communications (GSM) full rate (FR), GSM enhanced full rate (EFR) and 3GPP adaptive multi-rate (AMR).

6.7 Transcoding/multiple encoding

Transcoding, according to [ITU-T V.152], in general is the translation from one type of encoded media format to another different media format (examples for media type "voice": ITU-T G.711 A-law to μ-law or vice versa, ITU-T G.711 codec to ITU-T G.726, etc.).

6.8 Packet loss concealment

Packet loss concealment (PLC) algorithms according to Appendix I of [ITU-T G.711], also known as frame erasure concealment algorithms, hide transmission losses in an audio system where the input signal is encoded and packetized at a transmitter, sent over a network, and received at a receiver that decodes the packet and plays out the output. Many of the standard code excited linear prediction (CELP)-based speech coders, such as those defined in [ITU-T G.723.1], [ITU-T G.728] and [ITU-T G.729], have PLC algorithms built into their standards. The objective of PLC is to generate a synthetic speech signal to cover missing data (erasures) in a received bit stream. Ideally, the synthesized signal will have the same timbre and spectral characteristics as the missing signal, and will not create unnatural artifacts. Since speech signals are often locally stationary, it is possible to use the signals' past history to generate a reasonable approximation to the missing segment. If the erasures are not too long, and the erasure does not land in a region where the signal is rapidly changing, the erasures may be inaudible after concealment.

6.9 Voice activity detection

Voice activity detection is a function, which enables differentiating between wanted and unwanted inband voiceband signals, e.g., to obtain trunking efficiency in circuit multiplication equipment (CME), to ensure correct operation of echo control and other signal enhancement devices etc. (see also [ITU-T G.720.1]).

6.10 Signal type classification

Signal type classification is a process in which relevant characteristics from a transmission signal are extracted and used to identify into which of a set of classes the signal is most likely to fit – for example: speech, modem signals, fax-signals, dual-tone multi-frequency (DTMF), and signalling tones.

6.11 Digital speech interpolation

Digital speech interpolation (DSI), also called silence elimination, is a mechanism which takes advantage of inactive periods in a conversation to insert speech from other conversations and reduces the transmission capacity needed to handle a multiplicity of telephone trunk channels by exploiting the low average channel activity by transmitting active speech.

6.12 Tones/signal generation/detection

Tones/signal generation/detection is a generic mechanism to generate/detect tones/signals such as multi-frequency tones (R1/R2-Signalling) and DTMF.

6.13 FAX demodulation/remodulation

Fax demodulation/remodulation is a process of detecting facsimile traffic, extracting digital information from the incoming modulated signal, transporting this information across a transmission path and reproducing the facsimile control and image information by remodulation at the other end. Examples of implementations are found in digital circuit multiplication equipment (DCME) [ITU-T G.766] and the transmission of Group 3 facsimile from public switched telephone network (PSTN) or integrated services digital network (ISDN) over packet networks as defined in [ITU-T T.38].

6.14 Voiceband-data low rate encoding/decoding

A voiceband signal encoding method, e.g., adaptive differential pulse code modulation (ADPCM), which results in encoding, transmission and decoding of voiceband data signals. The purpose of this function is to reduce the size of voiceband data signal samples and frames in order to reduce the bitrate.

6.15 Data modem relay

Procedures that enable the transmission of Group 3 facsimile from PSTN or ISDN over the packet network e.g., Internet are defined in [ITU-T V.152].

6.16 Circuit-mode bearer services

Circuit-mode bearer service categories are defined in [ITU-T I.231.x] series.

6.16.1 64 kbit/s unrestricted digital information

Data transmission with 64 kbit/s (also called "clear channel" transmission) used e.g., for Group 4 fax equipment, video and dedicated signalling channels [ITU-T Q.700]. 64 kbit/s "clear channel" capability may be allocated for pre-assigned channels, dynamically requested channels (e.g., [ITU-T Q.50] protocol) or to voiceband data-low rate encoding/decoding (VBD) calls.

6.16.2 ISDN multi-rate bearer services

A point-to-point connection in which each endpoint, e.g., terminals, private automatic branch exchanges (PABXs) are communicating via the ISDN and using digital signals over multiple 64 kbit/s channels, in both directions simultaneously and continuously for the duration of the call. ISDN multi-rate bearer services are used e.g., for leased line services, ITU-T X.25 data services, private branch exchange (PBX)-to-PBX conferencing systems.

7 Signal processing network equipment

7.1 Type of signal processing equipment

7.1.1 Network EC

A network EC is a voice-operated device placed in the 4-wire portion of a circuit and used for reducing the cancelled-end echo present on the send path by subtracting an estimation of that echo from the cancelled-end echo (see Figure 1). The cancelled-end echoes consist of reflected signals caused e.g., at a 2-wire to 4-wire interface (hybrid).

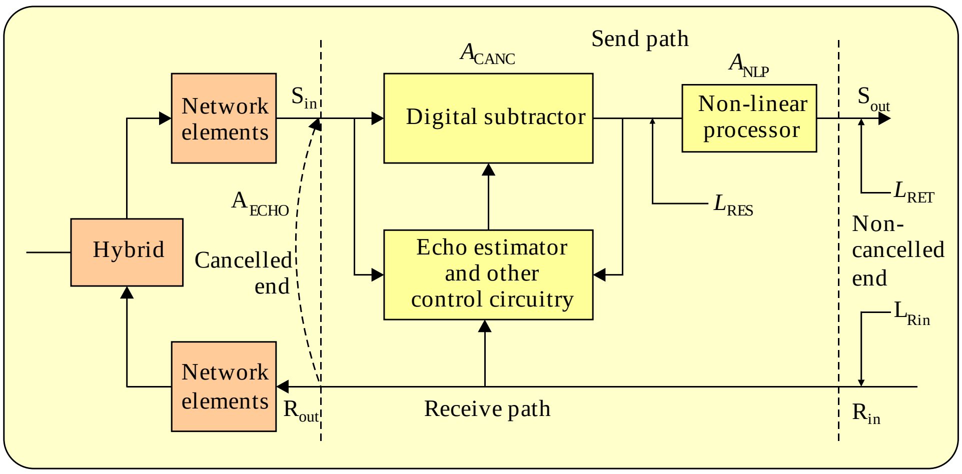

Figure 1 – Block diagram of an EC (ITU-T G.168)

A network EC according to [ITU-T G.168] may include the following functions:

– echo cancellation operation of up to an echo path capacity of Δ ms (e.g., 64 ms);
– NLP with comfort noise insertion;
– support of fax/modem ITU-T G.165/ITU-T G.168 tone disabling;
– support of signalling systems e.g., channel associated signalling (CAS), SS#5/#7, R2D, ITU-T Q.52, ITU-T Q.55, ITU-T Q.56, ITU-T Q.115;
– transparent digital data transmission (64 kbit/s channels) on selected time slots;
– remote/local monitoring and control;
– network interfaces e.g., 2 Mbit/s;
– redundant power supply (optional).

Application areas for network ECs (see also clause 7.2):

– digital wireless or cordless systems (e.g., in base stations of time division multiple access (TDMA), code division multiple access (CDMA), GSM networks);
– satellite communications and multiplexers (e.g., CME);
– wireline communications in PSTN/ISDN (e.g., in PBX and central office systems and used as a common equipment/EC pool) with long-haul connections/long propagation delays;
– voice over ATM, frame relay, Internet/LAN or packet switching systems and fax transmissions (e.g., gateways) with long-haul connections/long propagation delays.

7.1.2 Network AECs

Network AECs are devices placed in the 4-wire portion of a circuit and used for reducing cancelled-end acoustic echo.

7.1.3 Automatic level control device

An ALC device is a signal processing function located in the digital transmission path that automatically adjusts the level of a signal towards a predetermined value. An ALC device is designed to process signals in one direction of transmission (see Figure 2).

NOTE – For the purposes of this Recommendation, devices that modify the frequency response or spectral content of the signal in such a way as to affect the overall level of the signal are also defined as ALC devices.

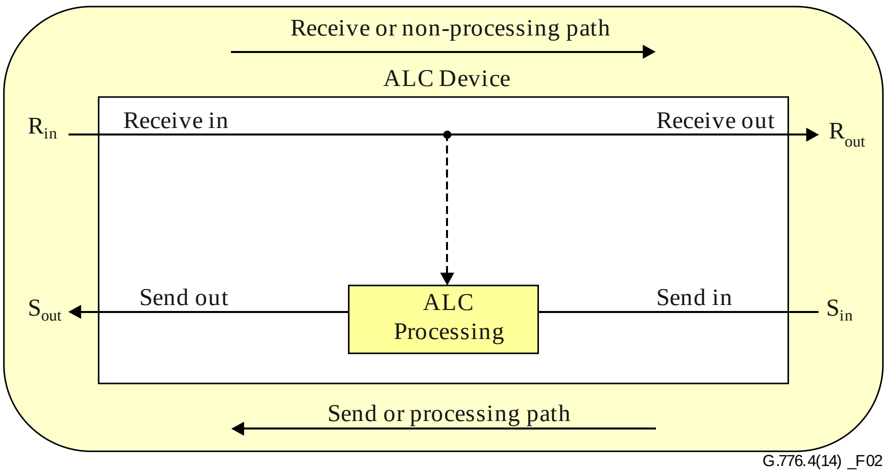

Figure 2 – Block diagram of an ALC device (ITU-T G.169)

An ALC device according to [ITU-T G.169] may include the following functions:

– adjustable gain/loss settings on all voice channelsAdjusts and optimizes the voice, transmit and receive levels;

– support of fax/modem ITU-T G.164 tone disabling;
– support of signalling systems e.g., CAS, SS#5/#7, R2D, ITU-T Q.52, ITU-T Q.55, ITU-T Q.56, ITU-T Q.115;
– transparent digital data transmission (64 kbit/s channels) on selected time slots;
– remote/local monitoring and control;
– redundant power supply (optional).

Application areas for ALC devices (see also clause 7.2) may include the following areas:

– digital wireless or cordless systems (e.g., in base stations of TDMA, CDMA, GSM networks);
– satellite communications and multiplexers (e.g., CME);
– wireline communications (e.g., in PBX and central office systems and used as a common equipment/EC pool) with long-haul connections/long propagation delays;
– voice over ATM, frame relay, Internet/LAN or packet switching systems and fax transmissions (e.g., gateways) with long-haul connections/long propagation delays.

7.1.4 Voice enhancement device

A voice enhancement device (VED) is defined as certain signal processing network functions (SPNFs) such as NR and AEC (see [ITU-T G.160], [ITU-T G.168] and [ITU-T G.169]) in the digital transmission path that perform voice enhancement functions on voiceband signals (see Figure 3).

Voice enhancement functions in a mobile network environment include the control of acoustic echo generated by wireless handsets, NR, and the recognition and accommodation of tandem free operation (TFO) and interworking function (IWF) signals.

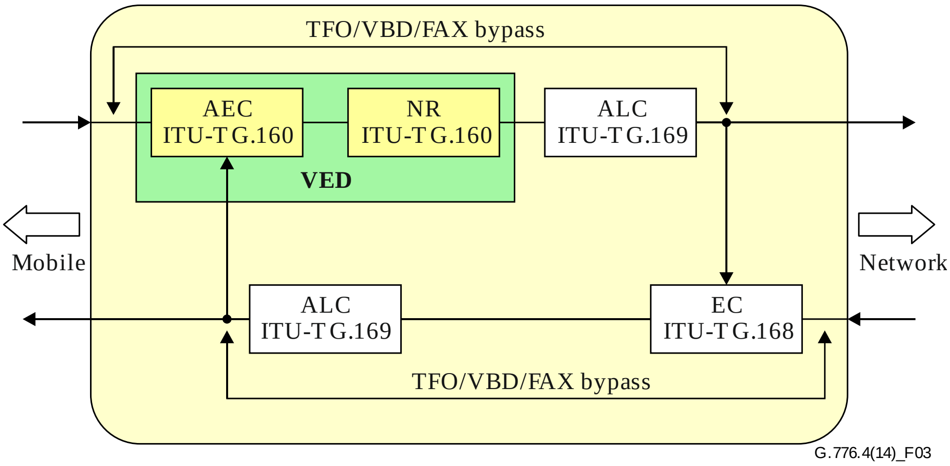

Figure 3 – Block diagram of a VED (ITU-T G.160)

An ITU-T G.160 VED may include the following functions:

– AEC operation with comfort noise insertion;
– NR operation;
– support of fax/modem ITU-T G.164/G.168 tone disabling;
– support of signalling systems e.g., CAS, SS#5/#6/#7, ITU-T Q.55, ITU-T Q.56, ITU-T Q.115, TFO;
– transparent digital data transmission (64 kbit/s channels) on selected time slots;
– network interfaces e.g., 2 Mbit/s;
– remote/local monitoring and control;

– redundant power supply (optional).

Application areas for VEDs (see also clause 7.2) are digital wireless or cordless systems (e.g., in base stations of TDMA, CDMA, GSM networks).

7.1.5 Circuit multiplication equipment

CME is a general class of equipment that permits concentration of a number of 64 kbit/s pulse code modulation (PCM) encoded input trunk channels on a reduced number of transmission channels in bearer channels (see Figure 4).

For the compression of speech and voiceband signals the following techniques can be used: DSI, speech low bitrate coding (ADPCM, LD-CELP, etc.), voiceband data-low rate encoding/decoding (VBD-LRE), Fax-demodulation/remodulation. The CME also provides 64 kbit/s assigned clear channels e.g., for unrestricted traffic. The operation modes of CME are contained in [ITU-T G.763], [ITU-T G.764], [ITU-T G.765], [ITU-T G.766], [ITU-T G.767] and [ITU-T G.768].

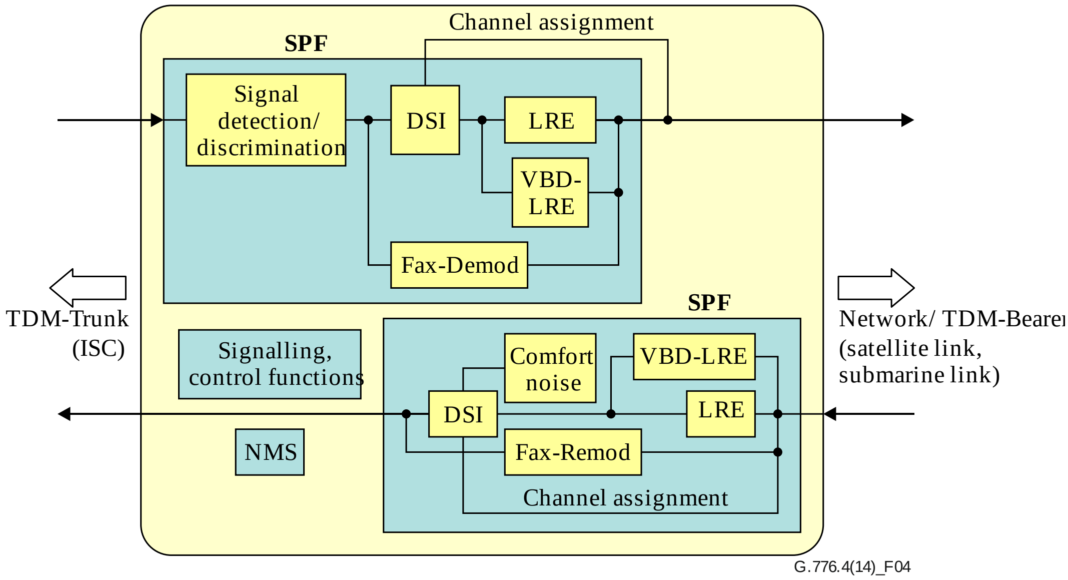

Figure 4 – Block diagram of DCME (ITU-T G.763 – ITU-T G.768)

The DCME may include the following functions:

– speech detection;

– interpolation of speech signals (DSI);

– transcoding of 64 kbit/s PCM to low bitrate coding such as ADPCM or LD-CELP;

– a function to detect and regenerate the background noise from the transmit direction;

– voiceband data detection;

– fax demodulation/remodulation for e.g., ITU-T V.17, ITU-T V.27, and ITU-T V.29;

– VBD-LRE;

– support of signalling systems e.g., CAS, SS#5/#6/#7, R1 and R2;

– International switching centre (ISC) data link interface in compliance with e.g., ITU-T Q.50, ITU-T Q.51;

– a function for exchanging signals with an ISC for purposes of ISDN bearer services involving 64 kbit/s unrestricted taffic [ITU-T Q.931];

– Network management system (NMS);

– system operation (capacity, overload strategy, maintenance, alarm); and

– one or more of the following operating modes:

i) point-to-point,

ii) multi-clique,

iii) multi-destination.

7.1.6 IP-based CME

Internet protocol circuit multiplication equipment (IP-CME) constitutes a general class of equipment that permits concentration of a number of IP ports on a reduced number of transmission channels over IP-based networks. IP-CME applies to digital circuit multiplication equipment optimized for IP-based networks (see Figure 5).

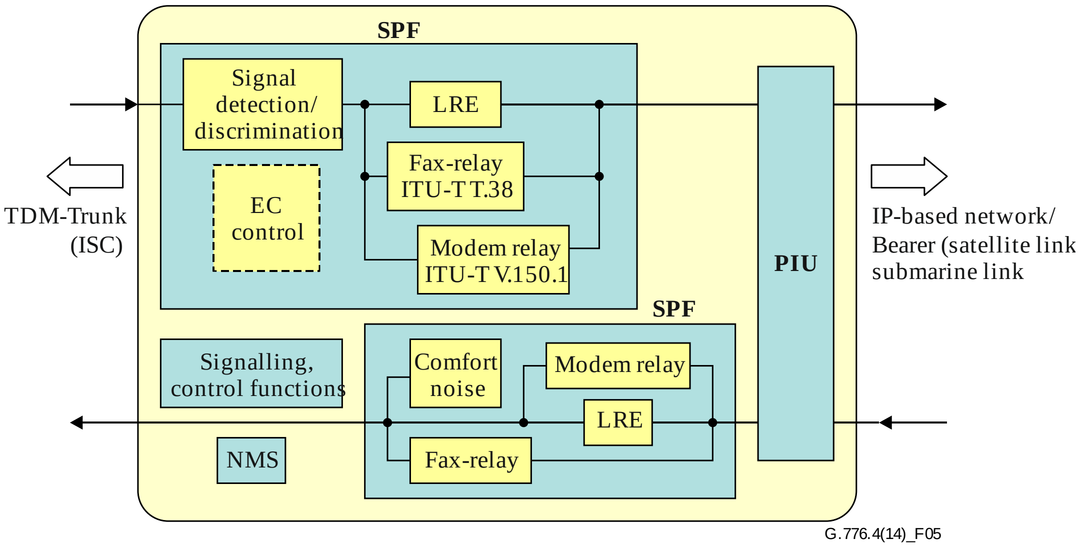

Figure 5 – Block diagram of IP-CME (ITU-T G.769/ITU-T Y.1242)

The IP-CME may include the following functions:

– speech detection;

– transcoding of 64 kbit/s PCM to low bitrate coding ITU-T G.723.1, ITU-T G.726 and ITU-T G.729;

– voiceband data detection;

– fax relay ITU-T T.38, modem relay ITU-T V.150.1;

– signalling detection;

– support of signalling systems e.g., CAS, ITU-T Q.931, SS#5/#6/#7, R1 and R2, SIGTRAN, IPv4/IPv6;

– ISC data link interface in compliance with e.g., ITU-T Q.50, ITU-T Q.50.1 and ITU-T Q.50.2

– multiplication schemes optimized for IP-based networks;

– handling of the IP-CME control signalling between IP-CMEs;

– multiplexing load control of IP-transmission channels over IP-based networks;

– NMS;

– system operation (capacity, overload strategy, maintenance, alarm);

– one or more of the following operating modes:

i) point-to-point,

ii) multi-clique.

7.1.7 Voice gateways

A voice gateway is a subset of a gateway that deals with voice and voiceband traffic only, and not data or video traffic (see Figure 6).

Voice gateways cover the requirements of equipment that interconnects goods and services tax networks (GSTNs) at the time division multiplex (TDM) interface and networks optimized for the transport of IP. [ITU-T G.799.1] describes functionality such as voiceband coding and echo control, transport interfaces, signalling interfaces, and OAM interfaces of these devices. Support is provided for calls from IP to GSTN, GSTN to IP, IP to IP and GSTN to GSTN. [ITU-T G.799.1] covers voice, voiceband data, facsimile, narrow-band digital data, address and signalling tones.

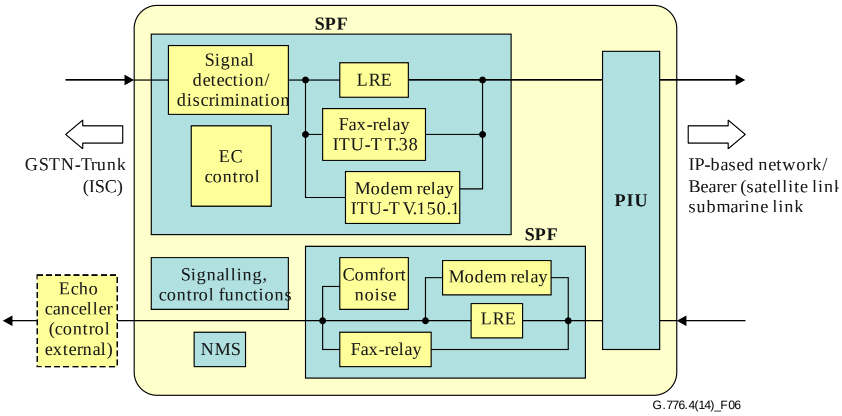

Figure 6 – Block diagram of voice gateway (TIGIN) (ITU-T G.799.1/ITU-T Y.1451.1)

The voice gateways may include the following functions:

– speech detection;

– interpolation of speech signals (DSI) internal or external to the codecs;

– transcoding of 64 kbit/s PCM to low bitrate coding e.g., [ITU-T G.723.1], [ITU-T G.726], [ITU-T G.728] and [ITU-T G.729];

– a function for transmitting detection and receiving injection of background noise, (including synchronous reset between encoder and coder);

– fax relay ITU-T T.38, modem relay ITU-T V.150.1;

– text telephony data relay ITU-T V.18;

– support of signalling systems e.g., CAS, ITU-T Q.931, SS#5/#6/#7, R1 and R2, ITU-T H.248/Megaco, SIGTRAN, IPv4;

– EC control data link interface in compliance with ITU-T Q.52/ITU-T Q.55;

– echo control logic ITU-T Q.115.1 (to media gateway controller via ITU-T H.248/Megaco messages);

– NMS;

– system operation (capacity, overload strategy, maintenance, alarm);

– one or more of the following operating modes:

    i) point-to-point,

    ii) multi-clique

NOTE – [ITU-T G.799.3] gateways are for further study.

7.2 Application areas of SPNF/SPNE

General application areas include:

– long distance telephony (e.g., echo cancellation and CME for satellite networks, long submarine cable connections);
– mobile/wireless/radio link networks (e.g., NR, echo cancellation devices);
– rural areas (e.g., CME for offshore oil rigs, distance learning, resorts);
– disaster relief (e.g., CME in multi-clique and multi-destination mode);
– civil engineering (e.g., CME, NR, echo cancellation).

The following figures cover a range of network scenarios where SPF/equipment are deployed.

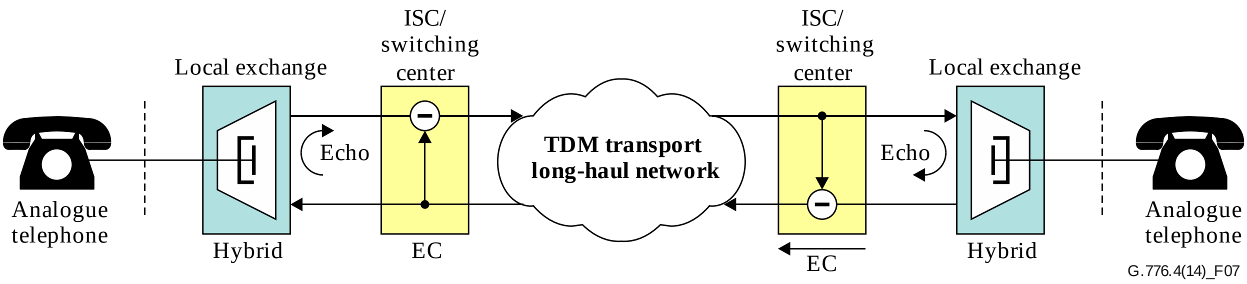

Figure 7 – International/domestic telephone calls with long-haul connections and analogue telephones

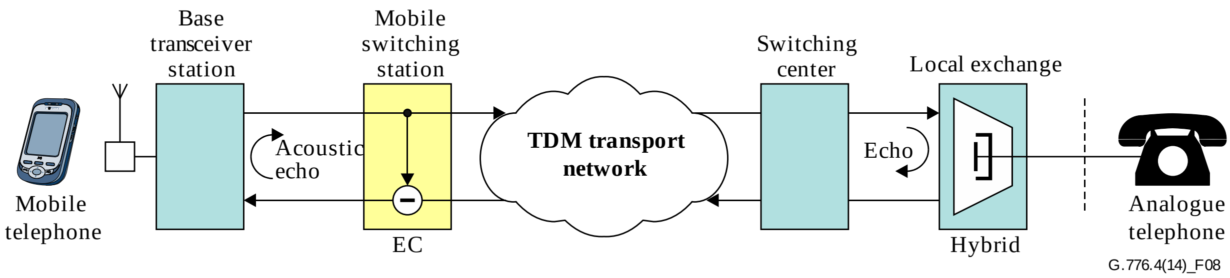

Figure 8 – Telephone calls between mobile and analogue telephones with EC

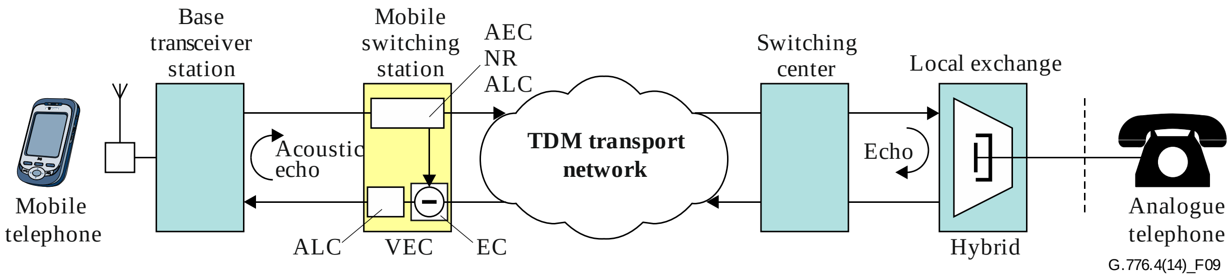

Figure 9 – Telephone calls between mobile and analogue telephones with VED

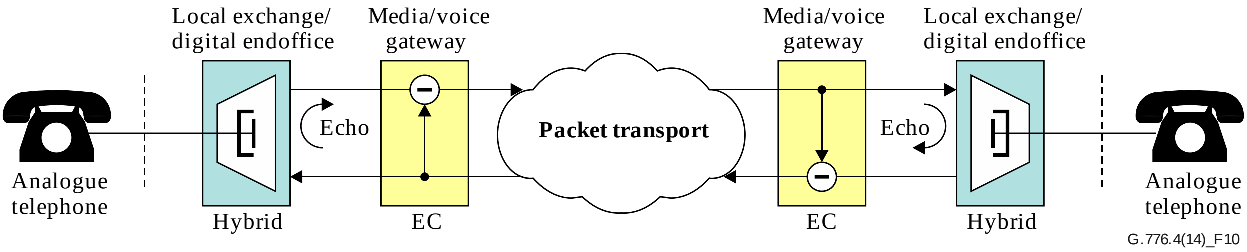

Figure 10 – Telephone calls between analogue telephones and media gateways in a TDM/Packet network environment

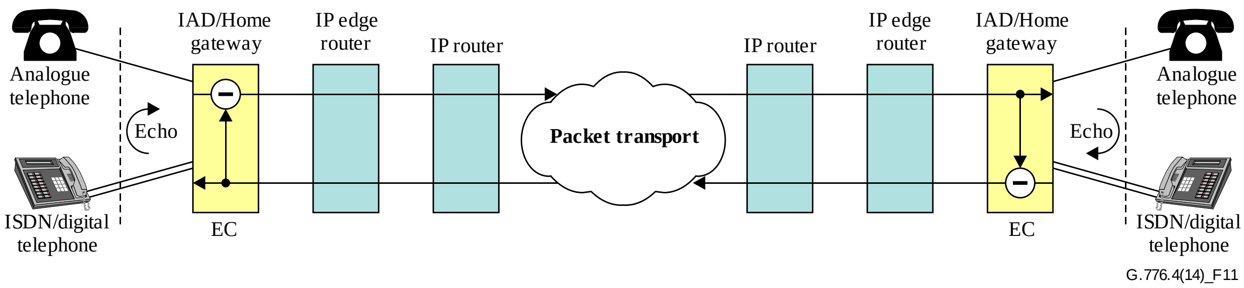

Figure 11 – Telephone calls between analogue/digital telephones and IADs/Home Gateways in an IP network environment

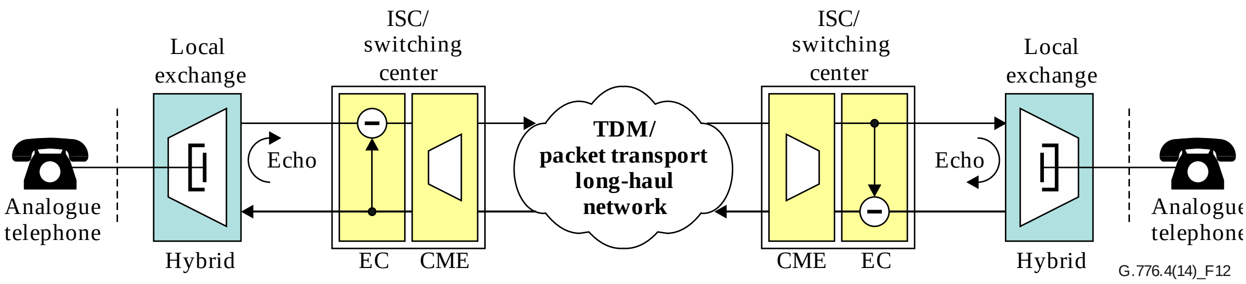

Figure 12 – Telephone calls between analogue telephones and CMEs

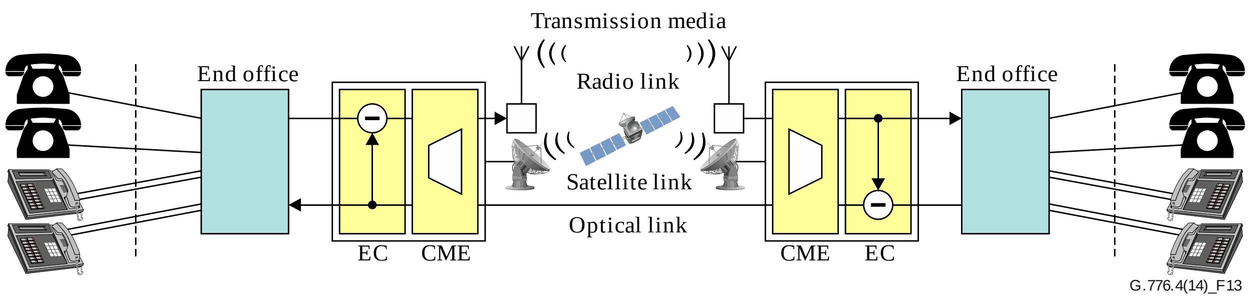

Figure 13 – Telephone calls over different transmission medias/applications and CMEs (including long distance/domestic telephony, cellular and call centre applications)

NOTE – All applications require echo path characteristics according to [ITU-T G.160] and [ITU-T G.168].

7.3 Delay requirements

Delay can be introduced from many sources in telecommunications networks and should be kept to a minimum. In PSTN/ISDN, delay is caused primarily by long-haul connections such as satellite or submarine sea cable connections. In wireless networks codec processing and air interface delays add significant delay so e.g., there is a high probability that a hybrid and/or acoustic echo in a mobile to PSTN/ISDN network connection can become noticeable. In voice over IP (VoIP) networks, buffering, packetization, codec processing and routing add an additional level of delay (see [ITU-T G.114] One-way transmission time). The degree to which talker echo becomes objectionable to the listener depends on the echo loudness and the delay between the original voice and the echo. [ITU-T G.131] (Talker echo and its control) presents results on the degree of annoyance of echo as a function of the amount of delay and talker echo loudness rating (TELR). According to this Recommendation for connections where the effects of talker echo are responsible for an undesirable decrease of transmission quality, the deployment of active echo control devices, such as ECs, is a valid choice. In general it is recommended to only deploy ECs into the network that conform [ITU-T G.168]. Echo suppressors according to [ITU-T G.164] and ECs according to [ITU-T G.165] may still be in use, but are not recommended for any new deployment. For details on transmission planning aspects of echo

ECs, see [ITU-T G.108.2] and [ITU-T G.161], and for details on EC control logic, see [ITU-T Q.115.0], [ITU-T Q.115.1], [ITU-T Q.115.2] and [ITU-T G.799.2].

8 Transport interfaces

8.1 TDM interfaces

If the SPNE has a TDM interface then it shall contain at least one of the interface types in Table 1. These interfaces may be also protected (e.g., 1+1 protection).

Table 1 – TDM trunk/bearer interfaces

Electrical	E1 E3 STM1	ITU-T G.703, ITU-T G.704 ITU-T G.703, ITU-T G.704 ITU-T G.707/Y.1322, ITU-T G.703	DS1 DS3 OC3	ANSI T1.102, ANSI T1.107
Optical	STM1 STM4 STM16 STM64	ITU-T G.707/Y.1322, ITU-T G.957 ITU-T G.707/Y.1322, ITU-T G.957 ITU-T G.707/Y.1322, ITU-T G.957 ITU-T G.707/Y.1322, ITU-T G.957	OC3 OC12 OC48 OC192	ANSI T1.105 ANSI T1.105 ANSI T1.105 ANSI T1.105

8.2 Ethernet interfaces

Examples of ethernet interfaces for SPNE can be found in Table 2. These interfaces may be also protected (e.g., 1+1 protection).

Table 2 – Ethernet bearer interfaces

Electrical	10/100 BaseT	IEEE 802.3 Clause 25
Optical	10/100 BaseFX	IEEE 802.3 Clause 26
Optical	1000 BaseX	IEEE 802.3 Clause 38
Electrical	1000 BaseT	IEEE 802.3 Clause 40
Optical	10 GBaseX	IEEE 802.3ae

9 Components and transmission characteristics

The following components can be provided in SPNE:

– audio codecs;

– echo cancellation;

– silence suppression;

– comfort noise generation;

– packet loss concealment;

– DTMF;

– jitter buffer.

Codecs

A SPNE may be provided with audio codecs such as [ITU-T G.711], [ITU-T G.723.1], [ITU-T G.726], [ITU-T G.728] and [ITU-T G.729]. In the case where two endpoints use different codecs by default, a negotiation process is necessary. A SPNE may also support transcoding if different codecs are used at each endpoint.

NEC

A SPNE should provide network echo cancellation according to [ITU-T G.168] if the mean one-way delay of the "talker echo transmission path" exceeds the limits contained in [ITU-T G.131] and [ITU-T G.114]. An echo path capacity can be selected which covers the echo path delay. The echo cancellation function can be provided e.g., for every port of a SPNE on a call-by-call basis. In particular, echo cancellation has to be switched off for transparent transport of transmission signals such as for clear channel mode (see clauses 6.15 and 6.16).

AEC

According to [ITU-T G.161] AECs can be installed either in the terminal itself (terminal AEC) or in places inside the network (network AEC).

Acoustic echo can be caused by acoustic coupling problems between a telephone's speaker and its microphone, e.g., in a hands-free set of a telephone and it can also be caused by crosstalk/sidetones in poor quality handsets or by echo in the environment surrounding the caller. Crosstalk/sidetones in handsets occurs when the sound coming out is coupled directly into the microphone.

Typical applications that may require AEC include:

– teleconferencing;
– loudspeaking (hands-free) telephones;
– videophone terminals;
– mobile handsets and personal applications.

Objective performance specifications for acoustic echo cancellation/controllers can be found e.g., in [ITU-T G.160], [ITU-T G.161] (clause 5.6), [ITU-T P.310], [ITU-T P.313], [ITU-T P.342].

Typical quantities that characterize the performance of acoustic echo cancellation/control are shown in the following list:

– echo loss in single and double talk modes;
– received and transmitted speech attenuations in double talk mode;
– received and transmitted speech distortions in double talk mode;
– break-in times in single and double talk modes;
– initial convergence time;
– recovery time after double talk;
– echo loss during echo path variations;
– recovery time after echo path variation.

Silence suppression and comfort noise insertion

During speech transmission, silent periods are detected. During the silent periods, speech transmission is suppressed and comfort noise can be inserted into the voice channel at the receiving side in order to prevent the impression that no information is sent during silent periods. For the comfort noise signal itself, in addition to the power level, the background frequency spectrum can be transferred. For music on hold, a special music detector may be implemented.

Packet loss concealment

PLC is a technique used to mask the effects of lost or discarded packets at the receiving end due to network delay, network congestion and/or network errors. Since the voice transmission is a real-time process, the receiver cannot request retransmission of the missing packets. Concealment algorithms are used to replace these lost packets. For PCM, specifically the [ITU-T G.711] coder and decoder (codec), a technique based on waveform substitution is used. Embedded PLC algorithm can be found in some compressed codecs.

DTMF tone transport

DTMF tones may be transmitted inband or outband in a SPNE. An outband transmission can be provided if a codec, e.g., [ITU-T G.723.1] cannot transmit the dual tones. Inband transmission can be provided by codecs, e.g., [ITU-T G.711], [ITU-T G.726] or via a special detection and transfer mechanism that processes the DTMF tone in its original power levels and signal length.

Jitter buffer

A jitter buffer is designed to remove the effects of jitter from the decoded voice stream, buffering each arriving packet for a short interval before playing it out synchronously. A fixed jitter buffer maintains a constant size whereas an adaptive jitter buffer has the capability of adjusting its size dynamically in order to optimize the delay/discard trade off (see [b-ETSI TS 102 929]). The jitter buffer size may have a configurable upper limit of a specific time.

10 Signalling interfaces

10.1 TDM signalling interfaces

The TDM signalling interface of a SPNE may consist of the following TDM signalling types:

– Signalling #5 – [ITU-T Q.140] and [ITU-T Q.180];
– Signalling #6 – ITU-T Q.251-Q.300 series;
– Signalling # 7 – [ITU-T Q.700 series];
– 64 kbit/s unrestricted taffic [ITU-T Q.931];
– V5.1 interfaces;
– V5.2 interfaces;
– ISDN primary rate interface (PRI);
– E1/fractional E1 interfaces – [ITU-T X.21], [ITU-T V.36];
– R1 signalling system – ITU-T Q.310-Q.332 series;
– R2 signalling system – ITU-T Q.400-Q.490 series;
– CAS as per [ITU-T G.704].

10.2 ISC signalling I/F and control function

ISC signalling and control function are used for dynamic load control of calls on the PSTN side and control of echo cancelling/VED mechanism, see [ITU-T Q.50x] series and [ITU-T Q.115.x] series.

10.3 IP bearer/packet signalling interfaces

The SPNE may consist of one of the following IP bearer/packet signalling interfaces:

– H.248/Megaco;
– SIGTRAN, IPv4/IPv6;
– [b-IETF RFC 791], [b-IETF RFC 950], [b-IETF RFC 919].

11 Management interfaces

The network management interface of a SPNE may consist of the following components:

– Command line interface (CLI) (A CLI is a local or remote interface for the control of an operation system. The CLI is used for e.g., configuration and maintenance, software installation and basic configuration, software backup and restore.).

– Simple network management protocol (SNMP) interface as a main management interface with a SNMP agent (A SNMP agent is a software process that responds to queries using the simple network management protocol SNMP ([b-IETF RFC 1157]) to provide status and statistics about e.g., a SPNE).
  - SNMP operational modes;
  - SNMPv2c (IETF RFC 1901 – IETF RFC 1908) uses UDP transport and simple community-based security scheme;
  - SNMPv3 (IETF RFC 3411 – IETF RFC 3418) uses enhanced user security model (USM) and supports message integrity, authentication and encryption.
– Fast ethernet interface;
– RS 232 interface;
– 1+1 protection switching.

12 NMS

A SPNE contains a set of system management and operation and maintenance (OAM) tools and features:

Configuration management

Configuration management is the process of organizing and maintaining information about all the components of e.g., SPNE such as:

– TDM interface features such as line rates, frame formats;
– IP interface features;
– EC options;
– voice feature options;
– clock recovery.

Fault management

Fault management concerns detection, isolation, and correction of abnormal operations. Issues to be considered include the management of e.g., equipment faults, facility faults at TDM and IP sides.

Performance management

Performance management assesses its ability to carry out all activities by continuously collecting and analysing statistical data related to key functions. Performance management includes items such as facility and connection performance monitoring, congestion control, end-to-end coordination and traffic management.

Security management

Security management contains the transport layer and user account security mechanisms. Security management including key management may be supported if a SPNE supports e.g., on IP-based per call control and signalling interfaces.

The following list gives an overview about available relevant NMS Recommendations:

– [ITU-T M.3010]: Principles for a telecommunications management network;
– [ITU-T M.3100]: Generic network information model;
– [ITU-T M.3400]: TMN management functions;
– [ITU-T M.370x] series: Common management services – Protocol neutral requirements and analysis;

– [ITU-T X.721]: Information technology – Open Systems Interconnection – Structure of management information: Definition of management information;
– [ITU-T X.731]: Information technology – Open Systems Interconnection – Systems Management: State management function;
– [ITU-T X.733]: Information technology – Open Systems Interconnection – Systems Management: Alarm reporting function;
– [ITU-T X.734]: Information technology – Open Systems Interconnection – Systems Management: Event report management function;
– [ITU-T X.735]: Information technology – Open Systems Interconnection – Systems Management: Log control function;
– [ITU-T X.736]: Information technology – Open Systems Interconnection – Systems Management: Security alarm reporting function;
– [b-TMF 513]: Multi-Technology Network Management Business Agreement;
– [b-TMF 608]: Multi-Technology Network Management Information Agreement;
– [b-TMF 814]: Multi-Technology Network Management Solution Set.

12.1 Managed objects, attributes and notifications

The operational configuration and status of the SPNE is represented by a set of managed objects, which combine physical and logical resources. Each managed object contains multiple attributes that represent different properties of specific resource. Each attribute has access permissions that indicate whether a specific property may or may not be modified by the user. Examples of managed objects that represent resources are e.g., TDM/IP trunks, EC_processing_capability, DLC data high load threshold, SPNE clock.

NOTE – [ITU-T G.776.1] (Managed objects for signal processing network elements) identifies the information model for the operations and management of SPNE.

13 Control and coordination procedures

13.1 Internal control

A SPNE has an operational capability of handling inband signalling systems such as signalling systems #5, R1 or DTMF digits.

13.2 External control

A SPNE has an operational capability of handling outband signalling systems such as signalling systems Q.50, #6, #7, R1/R2, H.248/Megaco packages and Sigtran protocols.

A SPNE (CME) may include different modes of channel assignments:

– speech coding handling in bearer channels;
– voiceband data traffic handling with assigned speech codecs, 64 kbit/s clear channel;
– fax traffic handling with e.g., ITU-T V.34;
– 64 kbit/s clear channel.

13.3 Coordination procedures

Coordination procedures are defined in [ITU-T G.799.2]. This Recommendation contains a framework and methodology for a dynamic coordination mechanism intended to minimize undesirable interaction of voice processing functions present on bearer paths of a communication link for improving overall end-to-end voice quality. The SPFs for coordination could be located in GSTN, IP network and NGN including mobile or vehicle terminal devices. The SPFs for coordination currently include EC, AEC, NR, ALC and automatic listener enhancement.

Soft coordination of VQE functions are for further study.

14 Clock recovery and synchronization

A SPNE may be controlled by different clock sources depending on the specific network application e.g., in plesiochronous/synchronous networks or between TDM- and IP-networks. In a traditional TDM service network, both ends of the TDM connection must be synchronized. If they are not synchronized, frames are either dropped (to prevent a buffer overflow condition) or inserted (to prevent an underflow condition). In both cases connection quality and reliability is affected.

The following clock sources may be applied:

– clocks derived from the trunk signals (e.g., recovered receive line clock from E1 interface);

– clocks derived from the bearer signals;

– external clock (e.g., T4 clock output from SDH equipment [ITU-T G.813], GPS receiver [ITU-T G.811]);

– internal clock.

Under normal operation conditions, a SPNE will utilize the primary clock source. Upon alarm conditions, failure or unavailability of the primary clock source, automatic switchover to the configured secondary clock source will occur. If both primary and secondary are unavailable or detected as failed, then a switchover to an internal clock source may occur (see also [ITU-T G.812], [ITU-T G.813]). Elastic buffers on the trunk side enable plesiochronous buffering and wander filtering ([ITU-T G.823], [ITU-T G.824], and [ITU-T G.825]).

The following system clocks may be generated:

– trunk output clock;

– transmit clock;

– receive clock.

Encoder and decoder of a SPNE in one transmission direction shall be driven by one clock source.

Timing and synchronization aspects related to SPNE in packet-based networks are for further study.

14.1 Jitter buffer

Packets transmitted over the network arrive at the receiving side with variable delays because the packets can take different routes or the network load can change during transmission. The variance in inter-packet arrival times is called jitter. There are different measures available which can compensate this variable delay, e.g., jitter buffer with a fixed delay or with a dynamic adaptation of the size of the jitter buffer according to the network demands. During data transmission such as for fax or modem signals, it is recommended to set a fixed value for the jitter buffer, see also [b-ETSI TS 102 929].

15 Reliability and redundancy

In order to maintain or restore an acceptable level of performance during operations, e.g., that any hardware failure will result in a hitless recovery with no active calls being lost, various restoration techniques and preventive techniques may be applied such as 1+1 redundancy of modules/power supply/fans, hot swapping of cards, hitless software upgrade capabilities.

CME may offer (in addition to the operating mode point-to-point, with multi-clique, multi-bearer, combined multi-clique multi-bearer and multi-destination modes) link solutions that use e.g., satellite links, microwave links in case of natural calamities and man-made disasters.

Figures 14 and 15 show applications of the different CME operation modes:

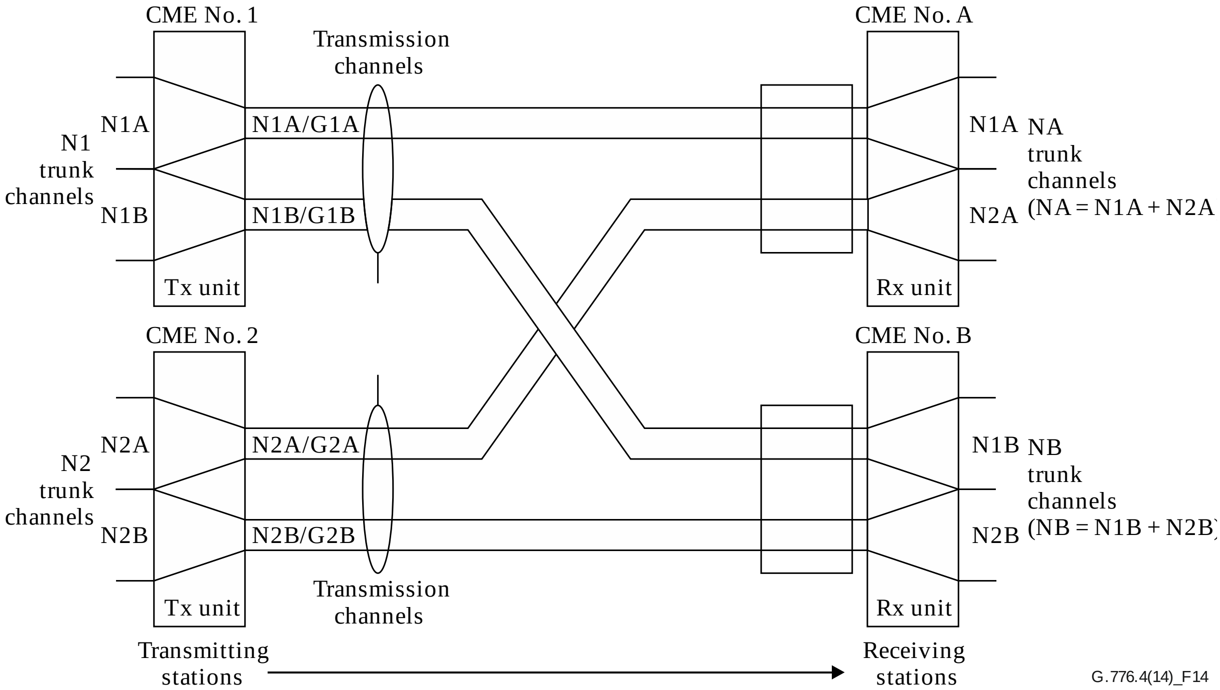

Figure 14 – Multi-clique mode (only one direction shown)

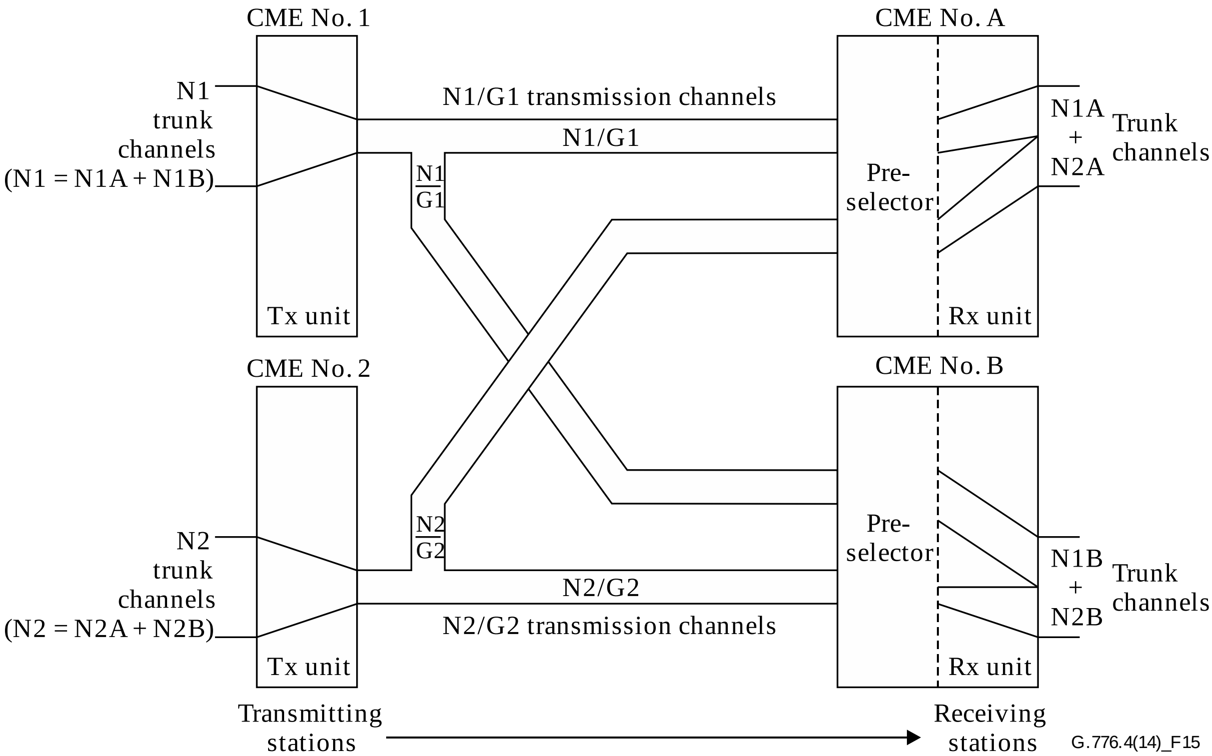

Figure 15 – Multi-destination mode (only one direction shown)

16 Environmental aspects of SPNE

Environmental aspects for SPNE are given in the following clauses.

16.1 Power supply

In addition to heat dissipation, power redundancy and number of power supplies, variables to consider include:

– operating input voltage range (within standard tolerances): e.g., −36 VDC to −75 VDC; 230 VAC;

– power consumption cards, shelf, rack e.g., 300 W, 1300 W (fully loaded);

– efficiency factor.

NOTE – By enhancing the efficiency of power and energy use to reduce consumption, the environmental load related to power generation, power transmission, greenhouse gas emissions etc. can be reduced (see also ITU-T L.1000-series).

16.2 Electromagnetic compatibility and emissions

Electromagnetic compatibility (EMC) is defined as the ability of equipment to function satisfactorily in its electromagnetic environment without introducing intolerable disturbances to anything in that environment. EMC requirements concern emissions and immunity.

The following compliances and requirements of standards may be applied:

International regulatory compliance

– EMC Emission e.g., CISPR, ITU-T K-series standards;

– EMC Immunity e.g., IEC, ITU-T K-series standards;

– Safety e.g., IEC, ITU-T K-series standards.

Continental/regional/national regulatory compliance

– EMC: ETSI EN standards;

– Safety: ETSI EN, CE standards.

Requirements regarding acoustic noise emission are shown e.g., in the standard [b-ETSI ETS 300 753].

16.3 Equipment/system dimensions and mechanics

An equipment/system may be housed in a rack cabinet and its units/terminals according to the following dimensions (see e.g., [b-ETSI ETS 300 119-4]):

– height: [mm], [in];

– width: [mm], [in];

– depth: [mm], [in];

– weight (fully configured): [kg], [lb].

16.4 Environmental conditions

An equipment/system may be designed for operation both at central offices and on customer premises. Compliance with relevant standards may be required. Relevant standards are, for example, the [b-ETSI EN 300 019-1-x] series.

For operation, storage and transportation the following items have to be considered:

Operation

– temperature range: [ºC], [ºF] (at x km/miles altitude);

–	humidity range: [%] RH (R.H. – relative humidity).

Storage and transportation

–	temperature range: [ºC], [ºF];

–	humidity range: [%] RH (R.H. – relative humidity).

### 16.5	Reliability and availability

Reliability is defined as the probability that a given equipment/system will perform its intended function for a given period of time under a given set of conditions. Requirements are specified using e.g., the following parameters:

–	mean time between failures (MTBF) / Failure in time (FIT), 1 FIT=10^(−9) 1/h);

–	mean time to repair (MTTR);

–	mean down time;

–	availability.

### 16.6	Test points

Test points are designed in general for easy and convenient accessibility during troubleshooting and should be protected against improper connection of e.g., test equipment.

### 16.7	Alarm interfaces

Alarm interfaces may consist of relay closures for visual and audio alarms and the alarm classification:

–	critical;

–	major;

–	minor;

–	warning;

–	information.

## 17	Conformance and interoperability

This Recommendation does not contain any conformance and interoperability requirements to be tested.

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