INTERNATIONAL TELECOMMUNICATION UNION

# ITU-T

# G.174

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

(06/94)

## GENERAL CHARACTERISTICS OF INTERNATIONAL TELEPHONE CONNECTIONS AND INTERNATIONAL TELEPHONE CIRCUITS

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# TRANSMISSION PERFORMANCE OBJECTIVES FOR TERRESTRIAL DIGITAL WIRELESS SYSTEMS USING PORTABLE TERMINALS TO ACCESS THE PSTN

## ITU-T Recommendation G.174

(Previously "CCITT Recommendation")

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# FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). The 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 Conference (WTSC), which meets every four years, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

ITU-T Recommendation G.174 was prepared by ITU-T Study Group 12 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 21th of June 1994.

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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## SUMMARY

This Recommendation provides transmission performance objectives that, if realized, should facilitate widespread user acceptance of emerging wireless technologies. These objectives apply to terrestrial digital wireless systems that use portable terminals to access the PSTN. Additionally, this Recommendation reinforces the fact that “comparable PSTN quality” encompasses, among many other considerations, a broad set of transmission performance criteria, all of which need to be considered to achieve the robustness and interworking capabilities of the PSTN.

## INTRODUCTION

Next generation wireless personal communication systems, such as the Future Public Land Mobile Telecommunication System (FPLMTS) being standardized by the ITU-R, are expected to be globally deployed by the year 2000. A clearly stated objective of FPLMTS is to have its quality of service, which includes transmission performance, be comparable to that of the PSTN. It is widely recognized that this objective should be satisfied in order for these wireless access systems to achieve the general utility and wide acceptance that has been realized by the PSTN. Accordingly, this recommendation was developed with the goal of providing transmission performance objectives that should be met by a wireless system in order to be considered comparable to the PSTN. These objectives were derived to be generic, and may or may not be directly applicable to FPLMTS, which may require other specific requirements.

Because wireless personal communication systems can in effect replace the national extension part of an international connection, all the Recommendations dealing with the national extensions can also be applied to these wireless systems, unless this Recommendation gives other guidelines for planning the system. The objectives given here were derived in recognition of the fact that they should be specific enough to meaningfully guide sub-system development, yet flexible enough to accommodate the well-established need to tradeoff between quality and capacity in a variety of realistic radio operating environments. Finally, this Recommendation highlights the diversity of transmission performance criteria that need to be considered in order for the “comparable to PSTN” quality goal to be realized.

# TRANSMISSION PERFORMANCE OBJECTIVES FOR TERRESTRIAL DIGITAL WIRELESS SYSTEMS USING PORTABLE TERMINALS TO ACCESS THE PSTN

*(Geneva, 1994)*

## 1 Scope

This Recommendation provides transmission performance objectives for terrestrial digital wireless systems that use portable terminals to access the PSTN (or the PSTN network interface).While satellite access to a base station is not within the scope of this Recommendation, interconnection with the PSTN, which may involve satellite access, or a satellite link within the PSTN, is not excluded. In this Recommendation, these portable terminals are generically referred to as wireless personal communication systems; however, there is no intent that the objectives herein necessarily apply to any specific wireless personal communication system. The objectives provided here have been derived with the explicit assumption that the switching and transmission systems used in the PSTN, above and including the local exchanges, are digital, with subscriber lines being either analogue or digital. This assumption allows for more pertinent guidance for the time frame in which these future wireless systems are expected to be deployed. Such wireless systems are not expected to introduce, during stable operation, any significant transmission quality degradation relative to PSTN links covered by the G.100-Series of Recommendations. Currently, Recommendation G.173 exists for established mobile technologies that may not comply with this Recommendation for economic reasons.

This Recommendation has two major purposes: first, to provide, in a single document, transmission performance objectives that, if realized, should facilitate widespread user acceptance of emerging wireless technologies. The second major purpose of this Recommendation is to reinforce the fact that “comparable PSTN quality” encompasses, among many other considerations, a broad set of transmission performance criteria, all of which need to be considered to achieve the robustness and interworking capabilities of the PSTN.

## 2 Normative References

| | |
|---|---|
| Recommendation E.430 | *Quality of Service Framework* |
| Recommendation E.800 | *Quality of Service and Dependability Vocabulary* |
| Recommendation G.113 | *Transmission Impairments* |
| Recommendation G.114 | *One-Way Transmission Time* |
| Recommendation G.131 | *Stability and Echo* |
| Recommendation G.165 | *Echo Cancellers* |
| Recommendation G.173 | *Transmission Planning Aspects of Speech Service in Digital Public Land Mobile Networks* |
| Recommendation G.711 | *Pulse Code Modulation (PCM) of Voice Frequencies* |
| Recommendation G.712 | *Transmission Performance Characteristics of Pulse Code Modulation* |
| Recommendation G.726 | *40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)* |
| Recommendation G.728 | *Coding of Speech at 16 kbit/s Using Low Delay-Code Excited Linear Prediction* |
| RecommendationP.31 | *Transmission Characteristics of Digital Telephone Sets* |
| Recommendation P.79 | *Determination of Loudness Ratings* |
| Recommendation Q.1001 | *General Aspects of Public Land Mobile Network* |

## 3 Abbreviations and Definitions

For the purposes of this Recommendation, the following abbreviations apply:

| | |
|---|---|
| ADPCM | Adaptive Differential Pulse Code Modulation |
| BER | Bit Error Ratio |
| BLER | Block Error Ratio |
| DTMF | Dual Tone Multiple Frequency |
| EPL | Echo Path Loss |
| FPLMTS | Future Public Land Mobile Telecommunication Systems |
| LD-CELP | Low Delay-Code Excited Linear Prediction |
| PCM | Pulse Code Modulation |
| PCS | Personal Communications Systems |
| PLMN | Public Land Mobile Network |
| PSTN | Public Switched Telephone Network |
| QDU | Quantization Distortion Unit |
| RLR | Receiving Loudness Rating |
| SLR | Sending Loudness Rating |
| $TCL_w$ | Terminal Coupling Loss (weighted) |

Where possible, definition sources have been included in brackets :

**base station**: The common name for all the radio equipment located at one and the same place used for serving one or several cells. [See Recommendation Q.1001.]

**handoff**: The action of switching a call in progress from one cell to another (or between radio channels in the same cell), to allow established calls to continue when mobile stations move from one cell to another (or as a method to minimize co-channel interference). [See Recommendation Q.1001.]

**public land mobile network (PLMN)**: A network established and operated by an Administration or Recognized Operating Agency (ROA) for the specific purpose of providing land mobile telecommunication services to the public. A PLMN may be regarded as an extension of a fixed network (e.g. PSTN) or as an integral part of the PSTN. [See Recommendation Q.1001.]

**PSTN**: In this Recommendation, this term is intended to represent a fully digital path in the switched network, local exchange to local exchange, that is terminated at the far end (from the wireless access system) by either an analogue subscriber line and terminal or by a digital line and terminal.

**wireless access**: Terminal access to the network using wireless technology. Examples are: digital mobile phones, digital cordless phones, and FPLMTS personal stations.

**wireline access**: Terminal access to the network which uses wireline technology (e.g. conventional telephone sets and subscriber lines).

**wireless terminal**: A general term used for any mobile station, personal station or personal terminal, with which non-fixed access to the network is used.

## 4 Reference Configuration

Figure 1 gives the general configuration of digital wireless personal access system to which the performance objectives of this Recommendation apply. When the network interface is digital, it should have a relative level of 0 dBr in both directions of transmission. When an analogue line is used to interconnect the base station to the digital paths in the PSTN, other considerations apply (see for example Figure 4).

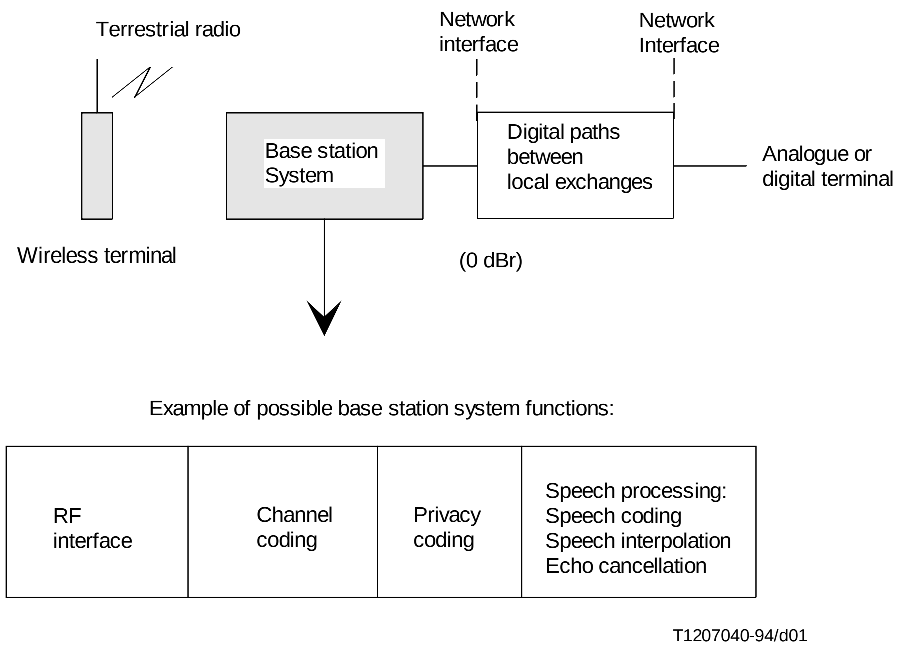


FIGURE 1/G.174

**Reference configuration for wireless access systems**

# 5 Overall PSTN Quality Perspective

## 5.1 PSTN Quality of Service Perspective

*Quality of Service* is defined by ITU-T Recommendation E.800 as the collective effect of service performances which determine the satisfaction of a user of a service. Quality of service is characterized by the combined aspects of performance factors applicable to all services, such as: service support performance; service operability performance; service accessibility performance; service retainability performance; service integrity performance; and other factors specific to a given service.

*Network performance* is defined as the ability of a network or network portion to provide the functions related to communications between users; it contributes to service accessibility, service retainability and service integrity. Network performance parameter values are usually derived from quality of service parameter values.

ITU-T Recommendation E.430 gives a high-level $3 \times 3$ matrix as the prime structure for identifying all quality of service sources and relationships relevant to network performance. This high-level matrix, summarized below in Table 1, recommends using the criteria of speed, accuracy and dependability to judge the quality with which the basic user functions of connection set-up, user information transfer, and connection release are performed. Developers of wireless personal communication systems are advised to refer to Recommendation E.430 for the list of specific ITU-T Recommendations that apply to each cell of this matrix. Careful consideration of this framework of telecommunication service quality should facilitate user acceptance.

TABLE 1/G.174

**High-level 3 × 3 matrix for quality of service**

| User functions | Quality criteria | | |
|---|---|---|---|
| | Speed | Accuracy | Dependability |
| Connection set-up | | | |
| User information transfer | | | |
| Connection release | | | |

## 5.2 PSTN Transmission Performance Perspective

This subclause gives high-level requirements that should be satisfied by wireless personal communications systems, if such systems are to be part of an end-to-end connection with transmission performance similar (in terms of user expectations) to that of the PSTN. Following the general descriptions given in this subclause, more specific transmission performance objectives are given. If these performance objectives are met, then it is highly likely that a level of performance would result that should satisfy the high-level requirements given now.

Connections including the wireless system should, under error-free conditions, achieve subjective ratings comparable to those of connections in the ordinary PSTN. This is a necessary, but in no way sufficient, condition for being considered “PSTN quality” or “toll quality.” To realize the widespread acceptance, from a quality perspective, of the PSTN, many other requirements should be satisfied by wireless systems. For example, the received speech should sound natural, and users should be able to recognize the voices of people with whom they are familiar. In addition, wireless systems should be robust to transcodings (such as when used in tandem with a far-end wireless system); robust to a reasonable level of bit and frame errors; and robust to the wide variety of ambient noise conditions (e.g. offices, outdoors, motorways) under which such systems will be used.

The list continues: highly interactive conversations should be possible with minor effort, meaning that excessive delays cannot be introduced; no annoying effects should be imposed on call progress tones, network announcements, or music-on-hold; severe channel impairments such as signal dropouts shall not be frequent or regular; speech processed through such systems shall be recognizable by network-based speech recognition systems (that already work well with PSTN-originated speech); voiceband data should be supported at a data rate and performance level expected by the user (to be quantified) for the mobile application they are using, which could be facsimile transmission or remote computer access (assuming these applications are features of the service being used); and of course in-band signals such as DTMF should be transmitted with a small probability of error by the DTMF receiver. (Voiceband data and signaling may not be supported in-band over the voice path, but by some other means.)

This subclause has highlighted some of the many expectations that users have of the PSTN. Failing to support these capabilities, or providing performance levels that prove unsatisfactory to the user, may cause wireless systems not to have the widespread user acceptance that is the clear goal for these systems.

# 6 Transmission Effects of the Digital Radio Channel

With wireless personal communications, one or more segments of the end-to-end path are carried over a wireless (radio) channel. This channel links a wireless terminal with a base station. In most wireless personal communications applications, it is unlikely that the wireless terminal will be within line-of-sight of the base station. Thus, a directly transmitted signal may be blocked by stationary and non-stationary structural and environmental objects such as walls, cars, and trees. Different propagation paths are produced by reflections from these objects, with each path having a different time delay, phase and attenuation. Due to signal additions from these different (multipath) propagation paths, a portable radio channel will experience signal fluctuations (i.e. multipath fading and dispersion) and distortions as a function of distance or time for a moving user. In fact, due to the motion of blocking objects in the vicinity of the user, even a stationary user will receive a signal that fluctuates with time. Interference from other users in adjacent cells may be experienced, particularly under fading conditions.

Additionally, different technologies (e.g. narrowband vs. wideband) will be affected differently by the various transmission effects mentioned. The system developer must take this into consideration in trying to satisfy the performance guidelines set forth in this Recommendation.

Figure 2 shows examples of received and detected signal envelopes as a function of time in the multipath environment. Fluctuations such as those shown occur in time (depending on the velocity of the user relative to the environment) and over distances of about a half wavelength (15 cm at 1 GHz). Fade and interfade durations depend on the velocity of the user, the carrier frequency, the fade threshold, channel bandwidth, and other factors. For narrowband systems, when the received signal fades below some noise-related or interference-related detection threshold (see Figure 2), an outage period occurs in which one or more frames can be lost, causing channel errors.

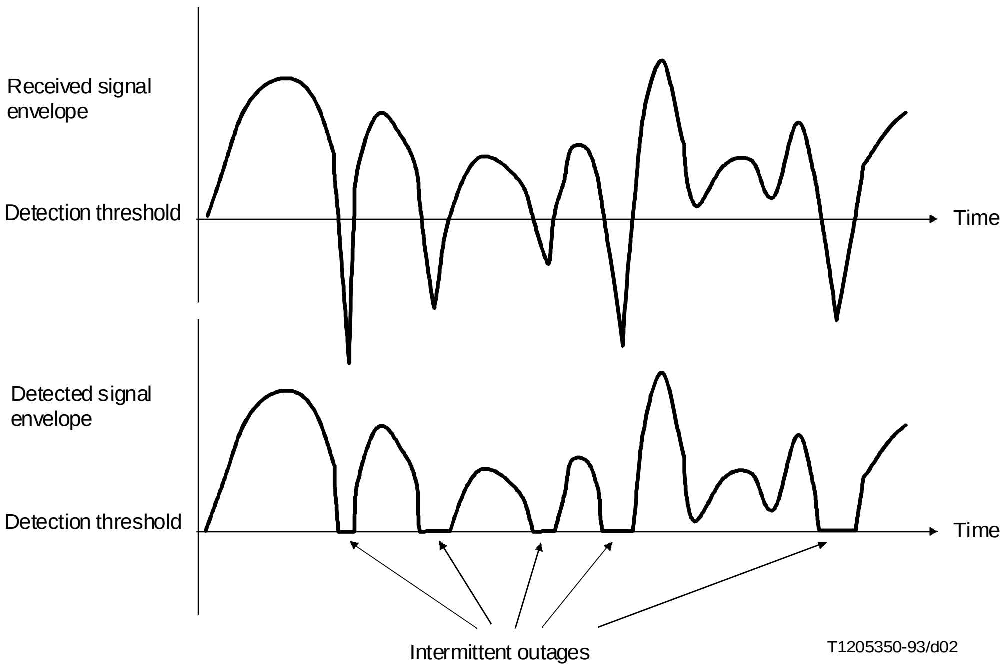


FIGURE 2/G.174

**Example of intermittent outages dues to fading (Narrowband system)**

# 7 Speech Transmission Performance Objectives

## 7.1 General

The transmission performance objectives that follow were derived to help future wireless systems meet the quality of service goal of being comparable to the PSTN.

For some transmission parameters, such as quantizing distortion, an “all-digital network” assumption will allow somewhat relaxed objective values for wireless access systems, in comparison to the more stringent values that would have been required for use in combination with a “hybrid” PSTN containing, for example, many analogue-to-digital conversions. It must be emphasized, however, that this trend does *not* apply to every transmission impairment. In particular, for the impairment of transmission delay, the “all-digital network” assumption leads to a *larger* value of this impairment in the network, due to the use of fiber, digital processing and buffering, and changes in network architectures and their associated routing. Extra scrutiny should therefore be exercised in considering various wireless system technologies.

## 7.2 Speech Coder Performance

As described above, the radio channel is characterized by relatively error-free transmission, punctuated by occasional bursts of lost frames during deep fading conditions. Thus, speech coders for wireless personal communications need to be evaluated in terms of their performance while being subjected to various lengths of correlated burst errors. Wireless personal communication systems are also likely to encounter higher ambient acoustic noise levels than wired communications. It is therefore also desirable that speech coders be evaluated in different amounts of acoustic background noise.

Of particular concern is the effect on the performance by radio channel fading while using low bit rate voice coders. In this case, personal communications users may experience previously unknown types of voice quality degradations resulting from the compression process itself, or from error multiplication when burst errors occur in the compressed data. The impact on perceived quality, due to burst errors on low bit rate speech coders, is an area for continuing study (see 7.11).

A useful test would characterize the speech quality of a coder under short, medium and long fading periods. Reasonable ranges for categories of fading could be: up to 25 ms (short), 25 to 75 ms (medium), and greater than 75 ms (long). Error bursts longer than about 100 ms introduce impairments independent of a coder's ability to recover, thus reducing the need for comparative testing for values > 100 ms. Speech coders that perform well for a representative range of short, medium and long error bursts should perform well in an actual wireless personal communications system.

Speech coders for wireless personal communications should be resilient when subjected to bursty errors, and should therefore have the following attributes:

– The speech coder should attempt to estimate the speech signal during short fading periods and during the initial segment (25 ms is suggested, based on the experience of some Administrations) of longer fading periods. No pops or clicks should occur.

– During longer fading periods, the output should decay without instability. The output should remain muted (or low level noise substituted) for the rest of the fading period.

– When a good frame is received after a fading period, the decoder should recover to the error-free output quickly, without resulting in any instability, pops or clicks.

Although it is not possible to recommend values for these attributes, they should be considered in the evaluation of candidate speech coders. Once an algorithm for simulating short, medium and long bursts of frame errors has been agreed on, all candidate coders should be subjectively tested using the same set of test sentences, and rated according to perceived speech quality. These tests will characterize how each coder performs under a wide range of burst errors, input levels, and tandeming (wireless-to-wireless interworking), so that the appropriate coder can be chosen based on the level of speech quality required for particular applications.

## 7.3 Loudness Ratings

Good wireless transmission performance should result with a nominal Sending Loudness Rating (SLR) of 8 dB, a nominal Receiving Loudness Rating (RLR) of 2 dB, nominal Sidetone Masking Ratings (STMR) of 10 to 15 dB, and a nominal Listener Sidetone Rating (LSTR) of not less than 15 dB. These values are consistent with ITU-T

Recommendation P.31, apply between the wireless terminal and the network interface (see Figure 1) and are typically intended for use in low ambient noise. Recommended values for high ambient noise are for further study (Annex E/P.79 may be useful). Additionally, a receive volume control may be desirable, with the specific range of operation being a subject for further study.

## 7.4 Weighted Terminal Coupling Loss

In order to provide sufficient echo protection (for the PSTN) due to acoustic coupling in the wireless terminal, a weighted terminal coupling loss ($TCL_w$) of at least 40 dB, and preferably 45 dB, should be provided (see ITU-T Recommendation P.31).

High acoustic isolation (> 40 dB $TCL_w$) may be achieved readily in standard handset terminals by careful design. However, in small portable or handsfree terminals, other more complex techniques may have to be used. For example, introduction of advanced echo control technology capable of increasing acoustic isolation in handsfree terminals may be needed (standard echo canceler technology is not likely to provide sufficient echo path loss enhancement in a dynamic acoustic environment). The echo control requirements of handsfree terminals are dependent upon the level of ambient noise; these requirements are under study.

## 7.5 Delay

Delay can have two effects on voice performance. First, it increases any echo impairment as perceived by users. Second, even when echo is controlled, one-way delays above 150 ms (end-to-end) can interfere (see Annex B of ITU-T Recommendation G.114) with the dynamics of voice conversation, depending upon the type of conversation and degree of interaction. In addition, delay can impair the performance of particular voiceband data applications, some being even more sensitive to delay than the voice application. These facts are pertinent for this Recommendation, whose purpose is to provide performance guidelines so that *terrestrial wireless access* systems achieve quality that is similar to that of the *wireline PSTN* segments that they replace (see the last paragraph of this subclause). It should be emphasized that Recommendation G.114 provides extensive guidance on transmission delay, including an acknowledgment that highly interactive user applications may be too restrictive for general network planning purposes, for which the one-way limit of 400 ms is used for complete (end-to-end) international connections

Echo control guidelines are necessary for *incremental* delay due to the wireless access portion of the connections to the PSTN. "Incremental delay" is the delay added by the wireless system above and beyond that of the wireline segment being replaced. It is recommended that the wireless access operator applies echo control measures when the one-way incremental delay is 5 ms or greater. (This value is based on planning guidelines commonly used by ITU-T Study Group 12.) This action will protect the wireless terminal user from annoying echo being returned from the far-end PSTN reflection, since it cannot be assumed that any PSTN trunks have echo cancelers (or adequate echo cancelers) on them.

For one-way incremental delays < 5 ms, careful assessment of potential echo-related degradation is strongly recommended. This consideration is mandated by the use of lightweight, pocket-sized handsets by some wireless systems in a variety of noisy environments, the combination of which could cause effects that are not completely understood. Thus it is recommended that careful testing be done.

*Total* one-way delay of wireless systems should still be limited even with the use of echo control. The reason is that these systems will be part of connections that may contain long fiber routes or satellites in the international PSTN segment. Although further studies are needed to provide specific delay and application trade-off guidelines for wireless systems, it is recommended to keep the total one-way delay below 40 ms, as was done for PLMN transmission planning in ITU-T Recommendation G.173. An objective value for one-way delay of less than 20 ms is desirable, given the ever-increasing concerns about the effects of total end-to-end transmission delay on user applications. (These delays refer to the one-way transmission times, which includes processing and propagation, between the wireless terminal and the base station.)

### 7.6 Delay and Echo Response Variations

Many wireless access systems may introduce switching delays and variations in echo response, such as those due to handoffs that suddenly change the transmission path. Delay variations especially affect the performance of modems with auto-ranging echo cancelers. Speech quality could also be degraded if network echo cancelers were affected. Accordingly, guidelines for limiting delay and echo response variation are desirable and are for further study.

### 7.7 Control of Echo from Outside the Wireless System

For 4-wire, all-digital connections, no echo control is needed beyond that of the terminal coupling loss given in 7.4. When interworking with PSTN digital paths that terminate in analogue lines, the wireless system needs to provide echo cancelers. To clarify this point, an example of echo generated at the far-end of the connection is shown in Figure 3. In this configuration an echo canceler is applied in the wireless access to control echo coming from the PSTN.

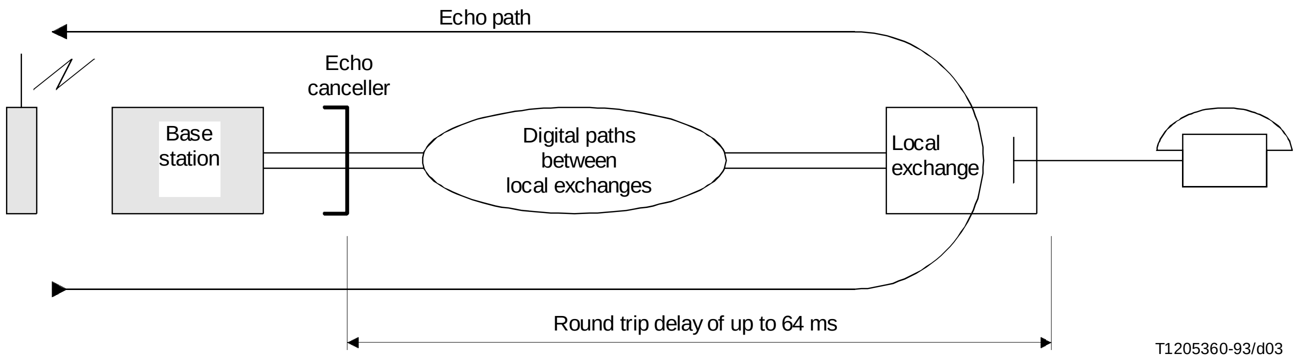


FIGURE 3/G.174

**Example of echo canceler deployment to control far-end echo**

Based on the existing PSTN infrastructure, echo cancelers deployed in wireless access systems should conform to ITU-T Recommendation G.165, and be able to handle up to 64 ms of echo path delay. It is likely that such echo cancelers will, in some cases, be working in tandem with PSTN echo control devices. This should not degrade the quality of the connection.

In some configurations, such as illustrated in Figure 4, near-end echo may be present together with the far-end echo. This may lead to additional performance degradation if any required echo control device is unable to cope effectively with both echo signals.

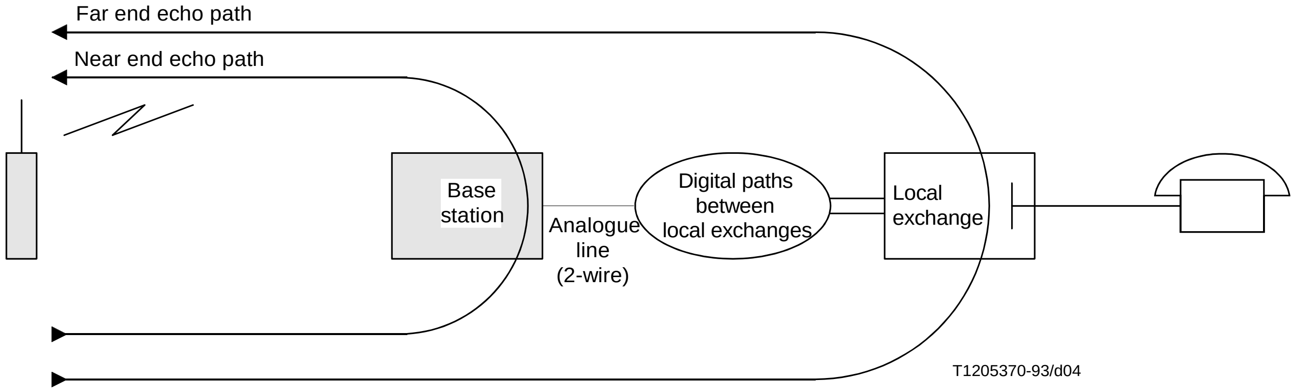


FIGURE 4/G.174

**Example of near-end echo and far-end echo**

## 7.8 Clipping (temporal)

Speech clipping is the loss of speech signal at any time, and can occur when, for example, speech interpolation is used, when low bit-rate coders change rate or during protection switching and uncontrolled slips. Clipping in this context does not refer to lost frames caused by burst errors in the radio channel. The subjective impact of clipping depends upon four factors: duration of clipping; percentage of speech clipped; frequency of clipping; and overall speech activity.

Based on the results of detailed subjective tests, and in consideration of a variety of transmission criteria practiced by the ITU-R and satellite consortia, two guidelines to maintain good speech quality are: clipping > 64 ms should always be avoided; and clipping < 64 ms should be kept below 0.2 percent of active speech. (Percent of clipped speech is 100 times the product of the frequency of speech clipping times clipping duration, divided by the speech activity factor.)

## 7.9 Idle Channel Noise

To be consistent with other ITU-T Recommendations, in particular G.712 and P.31, the following objectives are recommended:

With the input and output ports of the wireless system terminated in nominal impedance, the idle channel noise should not exceed –65 dBm0p (per Recommendation G.712). The level of any single frequency (in particular the sampling frequency and its multiples), measured selectively, should not exceed a level of –50 dBm0. Between 300 Hz and 3400Hz the level of any single frequency measured selectively and corrected by the psophometric weighting factor (see Table I/O.41) should not exceed –73 dBm0.

No existing ITU-T Recommendations address the noise requirements for digital wireless portable terminals; however, ITU-T Recommendation P.31 provides objectives for digital handset terminals, and these objectives may also be useful for wireless applications. Recommendation P.31 specifies –64 dBm0p in the sending direction (measured at the terminal interface) and –56 dBPa(A) in the receiving direction (measured using the artificial ear specified by Recommendation P.57, Type 1).

### 7.10 Noise Contrast and Comfort Noise

Noise contrast occurs when background noise is interrupted. This effect may be produced by several causes, such as echo cancellation using center clippers, speech interpolation, discontinuous transmission, etc. Comfort noise is noise that can be introduced to mask the negative effects of noise contrast. Different types of comfort noise may be used by different systems. Recommendations on noise contrast limits, and comfort noise types and values are for future study.

### 7.11 Random Bit Errors and Bursts of Errors

Good speech quality should be maintained during up to 3% frame erasures over any 10 second period (frame sizes on the order of 10-20 ms are assumed). The usual speech quality criterion is a reduction of no more than 0.5 mean opinion score unit (5 point scale) relative to the error-free condition.

### 7.12 Bandwidth

To maintain good speech quality and intelligibility, a passband of 300-3400 Hz (3 dB points) should be delivered, and sensitivity/frequency response should be in conformity with Recommendation P.31. For non-waveform coders, traditional measurement methods using single-frequency sine waves may not be adequate to evaluate effective bandwidth.

### 7.13 Stability Loss (Minimum Echo Path Loss)

For digital wireless access systems interfacing digitally to the PSTN, a minimum loss of 6 dB is recommended between the digital input and output paths of the wireless system (i.e. the 0dBr point according to Figure 1), at all frequencies in the range of 0 to 4000 Hz, under the worst-case acoustic conditions (e.g. with the handset placed face down on a hard, flat surface).

### 7.14 Quantization Distortion

Quantization distortion is introduced when an analogue signal is encoded to and from a digital format. ITU-T Recommendation G.113 characterizes quantization distortion in terms of a quantization distortion unit (QDU) which is defined to be equivalent to the distortion produced by one average A/D-D/A 64 kbit/s µ or A-law PCM codec conforming to ITU-T Recommendation G.711, under error-free conditions. The objective is that the coder/decoder pair of the digital wireless system should not introduce a value of more than 4 QDUs, consistent with the long-term objective for PLMN, per Recommendation G.173. (Values of QDUs for standardized coders, which only apply under stationary, non-degraded conditions), are provided in Recommendation G.113.) It is recognized that in some cases, such as low bit rate non-waveform coders, the QDU measure may not be appropriate, and other methods of specification should be used. This issue is under study.

It is also recommended that two wireless systems in tandem be no worse than the equivalent of three 32 kbit/s ADPCM (see ITU-T Recommendation G.726) coders asynchronously tandemed; this Recommendation is made to preclude too rapid an accumulation of distortion, and to allow for the presence of an ADPCM encoding in the international link. Further study is needed to determine how the distortion of two coders in tandem is related to their QDU ratings.

## 8 Voiceband Data and Other Non-Speech Transmission

### 8.1 General

Performance requirements for voiceband data (VBD) applications using wireless access systems should be stated in terms of the equivalent transmission metrics defined at the analogue interfaces on a connection, such as intermodulation distortion, phase jitter, envelope delay distortion, etc. ITU-T Recommendation G.113 has a number of annexes on VBD transmission performance in terms of impairment parameters. Definitions and some typical values for VBD impairments as well as a discussion of 32 kbit/s ADPCM as related to VBD performance are contained.

## 8.2 Application Requirements

User applications must be considered to determine desirable network performance levels. It is possible to classify most applications in a few broad categories depending on their accuracy performance requirements. Required modem performance, as a function of application, maps into related network performance requirements. Table 2 gives a classification of applications by accuracy parameter and needed limit.

Typical applications (e.g. facsimile) need fairly low error ratios, and thus the performance limit is stated in terms of the BER parameter. Forgiving applications (e.g. protected bulk data using a 1000-bit block) need a block error ratio (BLER) limit, derived from throughput considerations.

TABLE 2/G.174

**Modem Accuracy Performance as a Function of Application**

| Application | Parameter | Limit |
|---|---|---|
| Typical | BER | $10^{-5}$ |
| Forgiving | 1000 bit BLER | $10^{-2}$ |

## 8.3 Performance Suitability

Table 3 displays estimated suitability, for typical applications, of standard voiceband data modulation methods and various signal coding schemes, some of which could be used in digital wireless access systems. It is absolutely critical to emphasize that radio channel impairments are *not* considered in the table, and further study is needed to quantify the effects of these impairments. A “Yes” or “No” entry in the table simply indicates whether, based on available evidence and documentation, a *single* encoding of a given signal coding type is likely to acceptably accommodate the modem type at its maximum data rate. By “acceptable,” the application considerations of 8.1 should be taken into account.

In addition to radio channel impairments not being reflected in this table, the effects of tandem encodings of any combination of these coders on modem transmission are not included, and should be studied further.

Note that in cases involving higher-speed VBD, no low-rate coding will meet application needs. For example, facsimile operating at 9.6 kbit/s will not operate acceptably if passed through a 16 kbit/s coder. However, it may be possible to get facsimile to work at a lower data rate over the same channel. On the other hand, increasing the bit-rate of the codec (and thereby increasing the corresponding radio channel bandwidth requirements) can lead to increased higher-speed VBD application success.

## 8.4 Interworking

Since it is likely that wireless terminal users will want to transmit image (e.g. facsimile) or numerical (e.g. portable computer) data, it may be useful to include a data transmission capability in the mobile and base terminals. If the data are higher-speed and can not be successfully transmitted through the voice codec, then the digital bits could be coupled directly to the radio function (through the privacy codec). Special protocols and error control techniques might be needed to achieve acceptable error rates. See possible arrangements shown in Figure 5. In order to accommodate wireless-to-PSTN data connections, interworking provisions would need to be made.

TABLE 3/G.174

**Estimated Performance Suitability of Coder Types for various Modems**

| Modem Type at maximum Data Rate | Signal Coding Method (all ITU-T Recommendations) | | | | | |
|---|---|---|---|---|---|---|
| | G.711<br>64 kbit/s | G.726<br>40 kbit/s | G.726(1)<br>32 kbit/s | G.728(2)<br>16 kbit/s | Draft(3)<br>8 kbit/s | Draft(3)<br>4 kbit/s |
| V.21<br>300 bit/s | Yes | Yes | Yes | Yes | FS | FS |
| V.22<br>1200 bit/s | Yes | Yes | Yes | Yes | FS | FS |
| V.22 *bis*<br>2400 bit/s | Yes | Yes | Yes | Yes | FS | No |
| V.27 *ter*(4)<br>4800 bit/s | Yes | Yes | Yes | FS | No | No |
| V.32<br>9600 bit/s | Yes | Yes | Yes | FS | No | No |
| V.29(4)<br>9600 bit/s | Yes | Yes | No | No | No | No |
| V.32 *bis*<br>14 400 bit/s | Yes | FS | No | No | No | No |
| V.17(4)<br>14 400 bit/s | Yes | FS | No | No | No | No |
| V.34(5)<br>28 800 bit/s | Yes | FS | No | No | No | No |

FS = Further study is needed

NOTES

1 Entries based on G.721 standardization.

2 Preliminary.

3 Currently under study by ITU-TS SG 15.

4 This modulation scheme is used by Group 3 facsimile.

5 Two-wire standard under development by ITU-TS SG 14.

## 8.5 DTMF

DTMF signals may be used from the wireless terminals to interact with various DTMF-based features such as remote message retrieval. Wireless systems should therefore support the ability to transmit DTMF reliably, which for the PSTN is typically less than one DTMF errored signal in $10^4$ (i.e. under error-free PSTN conditions, the DTMF receiver produces an incorrect digit).

## 8.6 Call Progress Signals

Call progress signals, such as audible ringback and busy, should not be seriously degraded by the wireless access system.

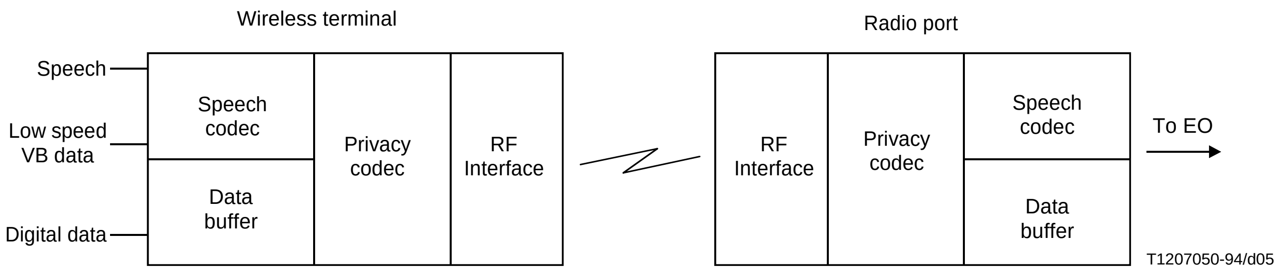


FIGURE 5/G.174

**Possible mobile terminal and base station arrangement for data transmission**