

Speech and multimedia Transmission Quality (STQ); Measurements of Call Establishment Performance in IP Networks



Reference

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Contents

| | |
|--|-----------|
| Intellectual Property Rights | 4 |
| Foreword..... | 4 |
| Introduction | 4 |
| 1 Scope | 5 |
| 2 References | 5 |
| 2.1 Normative references | 5 |
| 2.2 Informative references..... | 5 |
| 3 Abbreviations | 6 |
| 4 State of Art | 6 |
| 5 Need for a new indicator | 7 |
| 6 Measurements of call establishment performance | 7 |
| 6.1 Call Setup Delay [s] | 8 |
| 6.1.1 Abstract Definition | 8 |
| 6.1.2 Abstract Equation | 9 |
| 6.1.3 Trigger Points | 9 |
| 6.2 Media Establishment Delay [ms] | 9 |
| 6.2.1 Abstract Definition | 9 |
| 6.2.2 Abstract Equation | 9 |
| 6.2.3 Trigger Points | 10 |
| 6.2.4 Informer section..... | 10 |
| 6.2.5 Special Considerations..... | 10 |
| Annex A: Bibliography | 12 |
| History | 13 |

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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ).

Introduction

The existing definitions of Call Establishment Delay are related only to the context of telephony on traditional PSTN networks.

In VoIP context the Call Establishment may differ for several reasons. Firstly, the tones played by terminal do not necessarily correspond to the state of the network. For example, the user terminal may play the ring back tone even though it has not received any signalling message from the network informing that the called party has been alerted. In addition it can happen that some network resources are not available immediately after pick-up on caller part because the media channel needs to be established separately.

In these conditions it is necessary to define new terms characterizing the Call Establishment as well as the related measurement methodology.

1 Scope

The present document provides a state of art concerning definitions of call establishment delay existing in standardization. As call establishment in PSTN context and in IP context differ, the present document defines a new term of call establishment delay which better reflects the course of call establishment in VoIP networks. The call establishment delay can be reflected by two parameters: Call Setup Delay and Media Establishment Delay.

2 References

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Not applicable.

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area. For non-specific references, the latest version of the referenced document (including any amendments) applies.

- [i.1] ETSI EG 202 765-2: "Speech Processing, Transmission and Quality Aspects (STQ); QoS and network performance metrics and measurement methods; Part 2: Transmission Quality Indicator combining Voice Quality Metrics".
- [i.2] ITU-T Recommendation E.800: "Definitions of terms related to quality of service".
- [i.3] ITU-T Recommendation E.600: "Terms and Definitions of Traffic Engineering".
- [i.4] ITU-T Recommendation H.323: " Packet-based multimedia communications systems".
- [i.5] IETF RFC 3261: "SIP: Session Initiation Protocol".

3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

| | |
|-------|--|
| ETSI | European Telecommunications Standards Institute |
| IP | Internet Protocol |
| ISDN | Integrated Services Digital Network |
| ITU-T | International Telecommunication Union - Telecommunication standardization sector |
| PSTN | Public Switched Telephone Network |
| SIP | Session Initiation Protocol |
| TIA | Telecommunications & Information Administration |
| VoIP | Voice over Internet Protocol |

4 State of Art

Several definitions of call establishment delay are present in standards issued by different standardisation bodies: ETSI, ITU-T and TIA.

According to document EG 202 765-2 [i.1] the Post Dialling Delay *"is the time interval between the end of dialling by the caller and the reception back by him of the appropriate ringing tone or recorded announcement"*. This definition regards call establishment delay from end-user perspective.

ITU-T defines this parameter in two recommendations: ITU-T Recommendations E.600 [i.3] and E.800 [i.2].

The definition in ITU-T Recommendation E.600 [i.3] adds some significant adjustments to the definition by EG 202 765-2 [i.1]. In fact according to ITU-T Recommendation E.600 [i.3] the post-dialling delay is *"the time interval between the end of dialling by the user and the reception by him of the appropriate tone or recorded announcement, or the abandon of the call without tone"*. The fact of taking into consideration the abandon of the call without tone as an instant of call establishment termination can create some problems for supervision of service and for gathering of statistics concerning service functioning. For real calls this means taking into account the calls which have been terminated prematurely and for test calls (carried out by robots which terminate a call after timeout) it means taking into account calls which in fact were not successful.

ITU-T Recommendation E.600 [i.3] defines in addition the post-selection delay used in ISDN networks which is defined as follows:

"a) post-selection delay (overlap sending) is defined as the time interval from the instant the first bit of the INFORMATION message containing the last selection digit is passed by the calling terminal to the access signalling system until the last bit of the first message indicating call disposition is received by the calling terminal (ALERTING message in case of successful call).

b) post-selection delay (en-bloc sending) is defined as the time interval from the instant the first bit of the initial SETUP message containing all the selection digits is passed by the calling terminal to the access signalling system until the last bit of the first message indicating call disposition is received by the calling terminal (ALERTING message in case of successful call)."

It is worthy of noticing that ITU-T Recommendation E.600 [i.3] defines the event of call establishment in terms of protocol messages. It is the only definition which regards the call establishment delay from both perspectives: user and network.

In addition, one can also find a definition of call set-up time by ITU-T Recommendation E.800 [i.2]: *"The period starting when the address information required for setting up a call is received by the network (recognized on the calling user's access line) and finishing when the called party busy tone, or ringing tone or answer signal is received by the calling party (i.e., recognized on the calling user's access line). Local, national and service calls should be included, but calls to Other Licensed Operators should not, as a given operator cannot control the QoS delivered by another network."* This definition regards the call establishment delay from network perspective.

The last definition of call delay defined by TIA is based on document FS 1037C issued by US Federal Standards which defines the following: *"The time between the instant a system receives a call attempt and the instant of initiation of ringing at the call receiver end instrument"*. Yet again although named differently, this definition describes the call establishment delay from network perspective.

It can be noticed that the essence of all definitions is the same although they use different terms to describe it. There are some minor differences between these definitions, which generally stem from the fact that they consider the call establishment either from network or user point of view. In our opinion both approaches are valid. Call establishment delay considered from user perspective is generally called post dialling delay; whereas from network perspective different names are used: post-selection delay, call set-up time or call delay.

5 Need for a new indicator

The aforesaid discussion reflects well the call establishment delay definitions in PSTN and thus can be directly employed for measurements of this indicator. In PSTN networks the invitation to dialling signifies that the necessary resources to pass a call are available. In addition the ring back tone signal indicates that the call is established and that a circuit is reserved. The call can begin at any time which depends only on called party, as the network has prepared everything for the call to start.

The situation in IP networks is different. Not only the signal of invitation to dialling does not signify that the resources are available, also the ring back tone does not indicate that the network has prepared everything for the call to start. In fact, due to longer transmission delay in IP networks the call establishment for VoIP may take longer than in PSTN. Some devices are therefore inclined to diminish the waiting by groundlessly introducing ring back tone signal before the network has actually informed that the called party has been alerted. Therefore the fact of hearing the ring back tone signal in handset does not necessarily indicate that the telephone rings at the called party.

In addition, no matter which signalling protocol is used for call establishment, in VoIP telephony a media channel needs to be established separately. This may take less or more time. In some situations the interlocutors are obliged to wait for several milliseconds after the called party has answered the phone before they can actually hear themselves speaking, as the media channel is not yet ready. It seems natural to claim that the call has been established when the network has provided the means for both interlocutors to converse and hear one another. As this delay is usually perceived by users it should therefore be taken into consideration when defining call establishment indicators. Media establishment delay can be equal to zero if the media flow has been established before answering the phone. Notice that in situations where media flow establishment begins as soon as the called user is being alerted, this indicator may be user dependant. In fact, in such cases the more time it will take for the called party to answer the phone, the smaller this indicator may be. However this should not be seen as a drawback as the purpose of this indicator is mainly to evaluate user perception.

Consequently, it seems important to define a new term of call establishment delay which will better reflect the course of call establishment in VoIP networks. The call establishment delay can be reflected by two parameters: Call Setup Delay and Media Establishment Delay.

6 Measurements of call establishment performance

The measurements of call establishment performance in a passive way are done at one point while analysing the IP flows. The different phases of call establishment and associated signalling messages for SIP and H.323 have been presented on Figure 1.

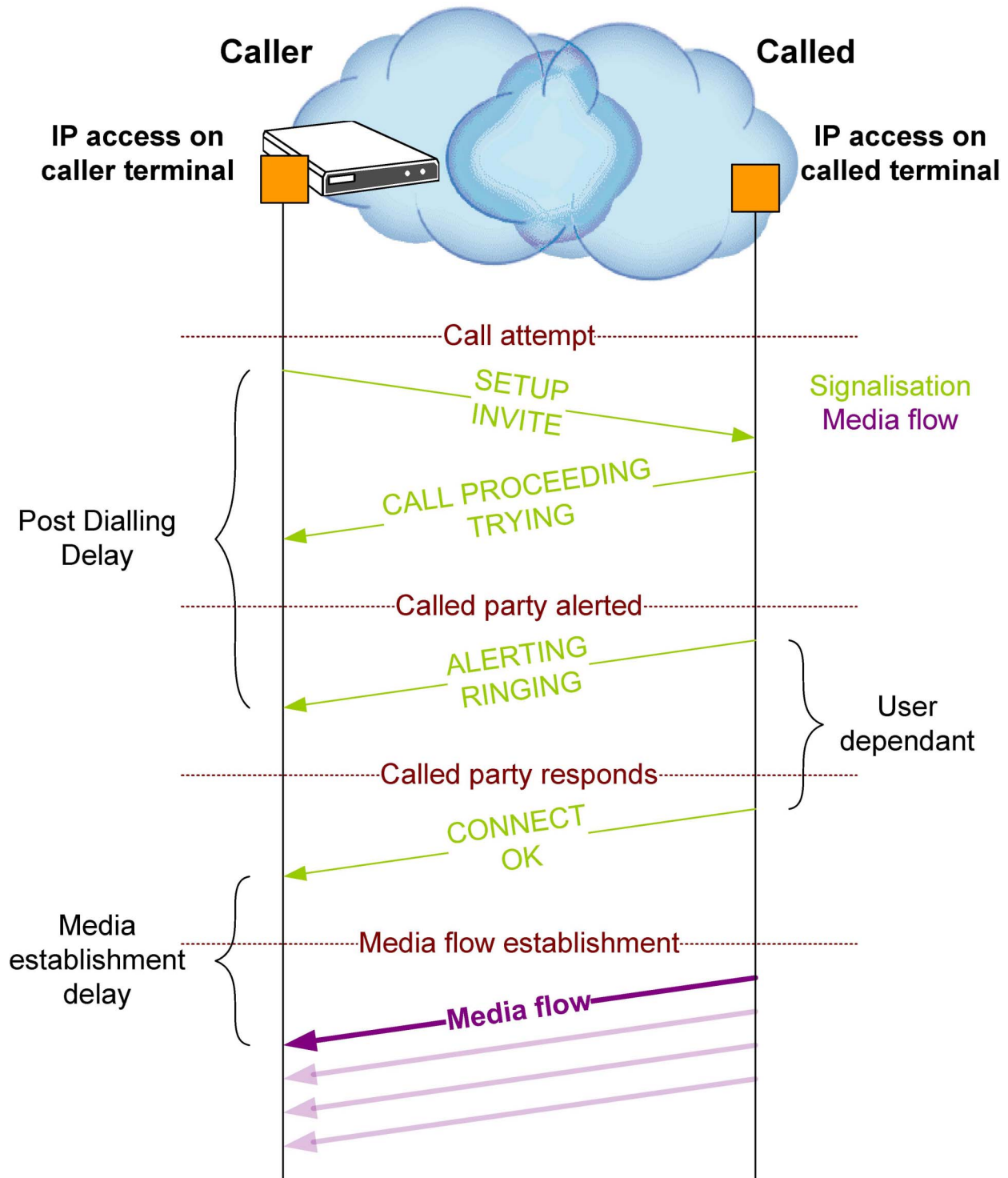


Figure 1: Call establishment phases in VoIP context

6.1 Call Setup Delay [s]

6.1.1 Abstract Definition

Time between sending of complete address information and receipt of call set-up notification.

6.1.2 Abstract Equation

$$\text{Call Setup Delay [s]} = (t_{\text{connect established}} - t_{\text{user presses send button on handset}}) [\text{s}]$$

NOTE: This parameter is not calculated unless the telephony call setup attempt is successful. Note that a call with "user busy" is also considered as a successful call setup.

6.1.3 Trigger Points

Table 1

| Event from abstract equation | Trigger point from user's point of view | Technical description/protocol part |
|---|---|--|
| $t_{\text{user presses send button on handset}}$: Time of call attempt | Start: Push send button (e.g. #) or end of timeout. (See note 3). | Start (H.323): The forming of first CONNECTION REQUEST message 'SETUP' Start (SIP): The forming of first CONNECTION REQUEST message 'INVITE' |
| $t_{\text{connection established}}$: Time when connection is established (successful call attempt) | Stop: Alerting tone is heard by the A-party coming from the B-party | Stop (H.323): The message informing of ring back tone 'ALERTING' or that the B-party user is occupied 'RELEASE COMPLETE' with release code 17 is received from the B-party. Stop (SIP): The message informing of ring back tone '180 RINGING' or that the B-party user is occupied '486 Busy Here' or '600 Busy Everywhere' is received from the B-party. |
| NOTE 1: With automatic tools there is not a significant difference between considering the alerting or the connect message, as the answer machine should always answer immediately. | | |
| NOTE 2: For the trigger points of the technical description/protocol part see Figure 1. | | |
| NOTE 3: For technical reasons if the push send button is not pressed by user telecom systems wait a timeout before processing the dialled number. This timeout is excluded from the measurement as it can be cut short by the user. | | |

6.2 Media Establishment Delay [ms]

6.2.1 Abstract Definition

Time between the moment the B-party answers the phone and the moment when media flows are established in both directions.

6.2.2 Abstract Equation

$$\text{Media Establishment delay [ms]} = \max((t_{\text{media flows established}} - t_{\text{B-party handset off-hook}}); 0) [\text{ms}]$$

NOTE 1: This parameter is not calculated unless the telephony call setup attempt is successful and the call was established (e.g. no busy tone).

NOTE 2: In some systems the media flow is established before the B-party handset is off-hook. In such situations the Media Establishment delay should be equal to zero (not negative).

6.2.3 Trigger Points

Table 2

| Event from abstract equation | Trigger point from user's point of view | Technical description/protocol part |
|---|--|---|
| $t_{\text{B-party handset off hook}}$: Time of taking the handset off-hook at the B-party | Start: B-party answers the phone and end of ring back tone at the A-party. | Start (H.323): The receiving of CONNECTION message 'CONNECT' Start (SIP): The receiving of CONNECTION message '200 OK' |
| $T_{\text{media flows established}}$: Time when media flows are established | Stop: A-party can hear B-party and inversely | Stop: The reception of first media packet. (See note 5). |
| <p>NOTE 1: With automatic tools there is not a significant difference between considering the alerting or the connect message, as the answer machine should always answer immediately. The user dependant time is therefore negligible.</p> <p>NOTE 2: This parameter cannot be evaluated at analog interface as the connection message is not received.</p> <p>NOTE 3: Even if there is a difference between the time the called user lifts the handset and the time a CONNECTION message (e.g. CONNECT for ITU-T Recommendation H.323 [i.4] or 200 OK for RFC 3261 [i.5] SIP) arrives at the caller side, this difference is small enough to allow the measurement of media establishment delay at the caller side in order to avoid expensive time synchronisation between the caller and the called side.</p> <p>NOTE 4: For the trigger points of the technical description/protocol part see Figure 1.</p> <p>NOTE 5: The first packet sent is not necessarily the first packet received. See considerations regarding NAT functions in clause 6.2.5.</p> | | |

6.2.4 Informer section

For analog terminals there exists a delay between the instant of taking the handset off-hook and the moment when the terminal is ready to transmit and receive speech. This delay is not embraced by Media Establishment Delay.

6.2.5 Special Considerations

Special caution needs to be taken if the network implements the NAT functions (e.g. in Session Border Controller or in Home Gateway). In fact NAT will drop all packets (usually around 6 packets to 10 packets) arriving from the B-party unless A-party also sent some media packets (Figure 2).

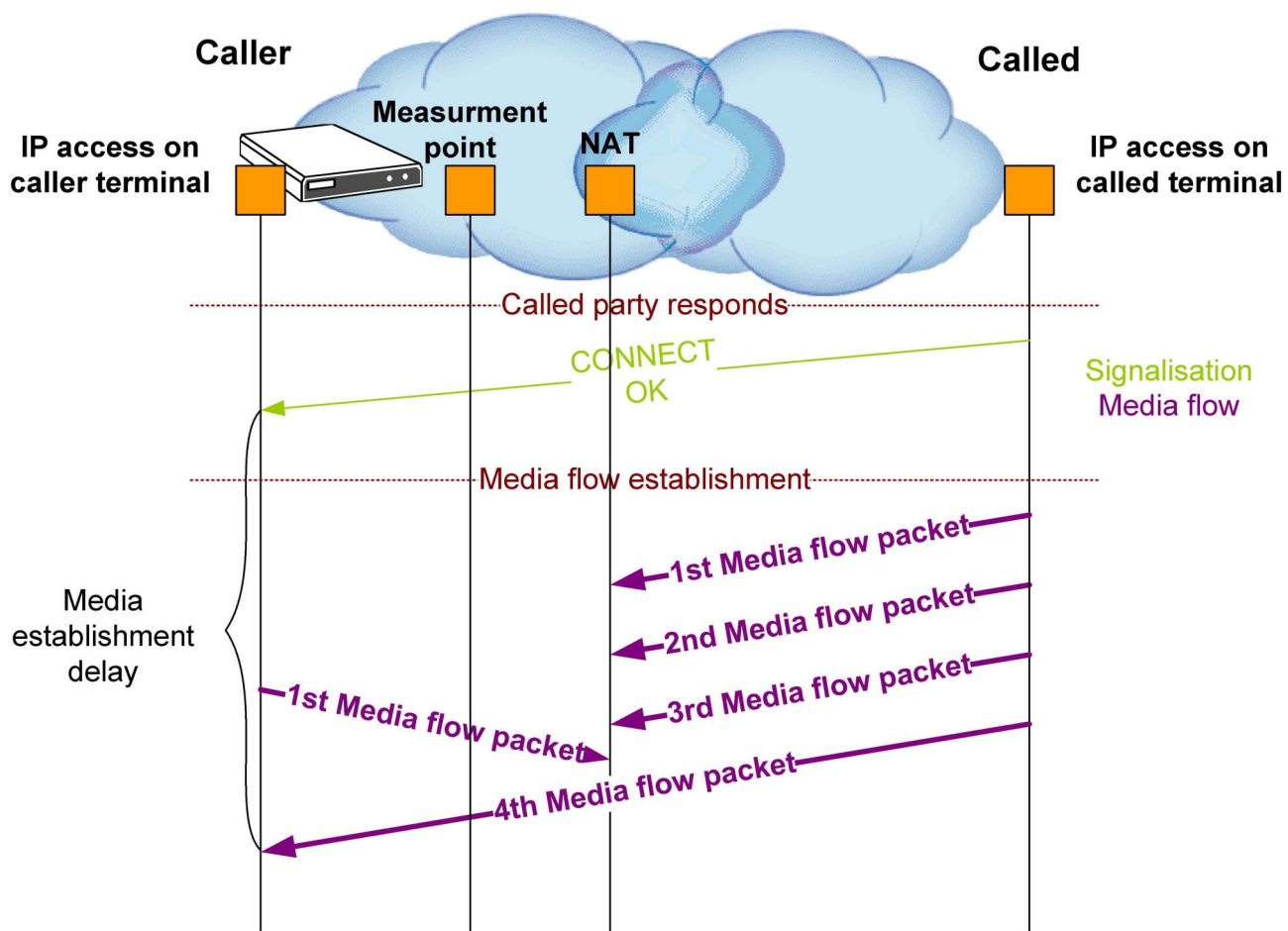


Figure 2: Dropping of media flow packets by NAT functionality

Measurement of media establishment delay can therefore take place in terminal or in network as long as the measurement takes place outside the NAT functions.

Annex A: Bibliography

TIA FS 1037C: "Telecommunications: Glossary of Telecommunication Terms".

History

| Document history | | |
|-------------------------|------------|-------------|
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