Broadband Telephony

Terminal Equipment Requirements and Specifications

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1 Introduction

1.1 Scope

This document provides a description of the Broadband Telephony (BBT) Service and lists the high level requirements and specifications which shall be supported by the terminal equipment in order to interoperate with Cyta's network infrastructure.

1.2 Intended audience

The intended audience of this document includes third party suppliers who would like to supply to the Cyprus market terminal equipment for the BBT service.

1.3 Notice

It should be emphasized that this document lists the high level requirements and specifications to which the terminal equipment shall conform to, in order to achieve interoperability with Cyta´s network infrastructure. This document, however, shall not be interpreted as a detailed specification document for third party suppliers in order to procure terminal equipment and cannot guarantee full interoperability of the terminal equipment with the core network. Cyta´s extended testing activities and experience is a proof that full interoperability can be only achieved on the field, mainly due to the specifics of the protocol implementations of the various vendors. It should be mentioned that this document does not include requirements for features which are network agnostic such as LCD dimensions and capabilities, address book capacity, ring tones etc.

1.4 Definitions

Client: A software application executing on a general-purpose Personal Computer. The Computer has IP access towards the Session Controller of the IP Multimedia Subsystem IMT3.0.

Terminal: A specific hardware device designed for Voice and/or Video telephony based on the SIP protocol. The device may be either a standalone unit or an adapter used to connect traditional POTS devices.

Intelligent Client: A client/terminal which can handle multiple calls without any assistance from the network, i.e., handle additional SIP invites while already involved in a session.

2 BBT Service and Network Infrastructure

2.1 BBT Service Description

The BBT service offers a package of voice services over a broadband connection, using the Voice over IP technology. Cyta´s subscribers, having a land line analog subscription and a DSL Access service, have the option to add additional IP voice lines with a maximum number of 4 VoIP lines (i.e. 1 existing POTS, 4 VoIP).

The BBT service users are able to:
- have up to five simultaneous voice conversations (1 POTS, 4 VoIP),
- have up to 4 voice conversations (1 POTS, 3 VoIP) together with 1 video conversation,
- use, in parallel, their existing DSL Access and Cytavision services.

The terminal equipment which can be used to support the service includes:
- Integrated Access Devices, i.e., xDSL modems equipped with additional fxs (Foreign Exchange Subscriber) interfaces and native SIP clients for establishing VoIP calls using plain analog phones.
- IPVideophones and/or IPphones with native SIP clients for establishing Video and/or Voice calls.
- Clients (Softphones), i.e., software applications which run on general purpose customer´s Personal Computers and which can be used for establishing Voice and Video calls with the use of headsets and cameras.
- Dual mode phones, i.e., GSM phones with WiFi capabilities and build-in SIP clients, which can be used to establish voice calls via WiFi access.

2.2 Devices tested and offered by Cyta

Cyta has successfully tested the interoperability of a number of devices which could be used for the BBT service and could be acquired by Cyta´s customers via the Cytashop.

These devices are listed below:
- Thomson Integrated Access Device, ST780 WL v.6.2.32.3468
- Grandstream IPVideophone, GXV-3000 v.1.0.1.20
- Siemens IPphone, Gigaset C470IP
- Movial Softphone, Cyta PC communicator v.6.4.28.2984
- Nokia dual mode mobile phones, N95, N81, E51, E65, E66

2.3 Network Infrastructure

The high level network infrastructure which has been deployed to support the BBT service is depicted in the following diagram. The core network consists of the IP Multimedia Subsystem (IMS) IMT 3.0 provided by Ericsson and the Softswitch Network (MGC hiE9200 S3.2, MG hiG 1000 v3.0T) provided by Nokia Siemens Networks.

Calls between VoIP subscribers are established solely utilizing the capabilities and resources of the IMS network. Calls originated from BBT subscribers towards PSTN/ISDN, GSM, and International destinations are routed via the IMS network to the SSW network which routes the call to the end customers´ Local Exchanges.
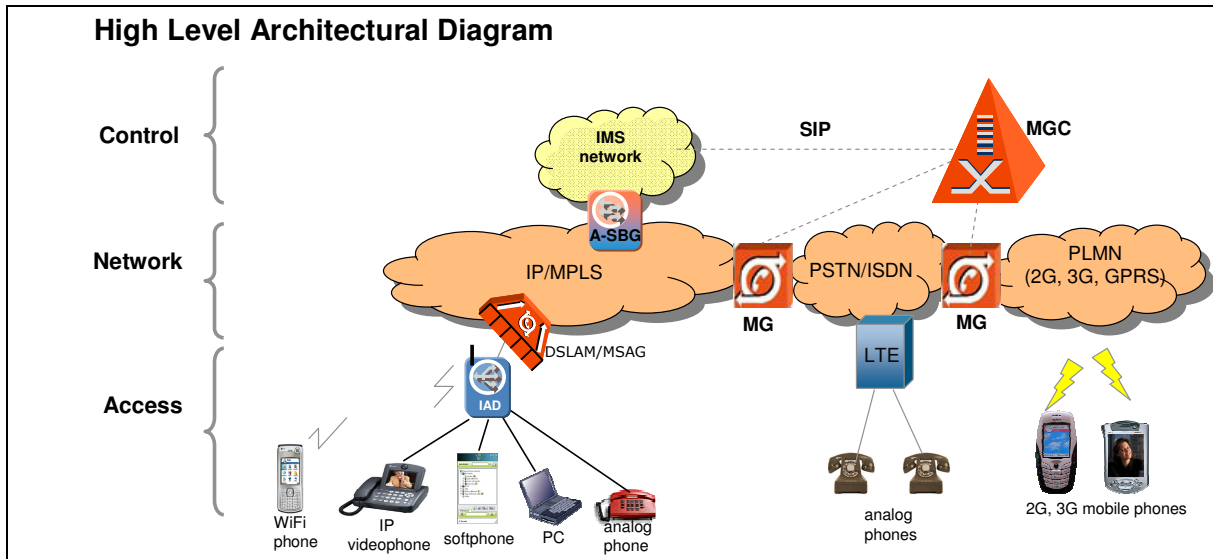
High Level Architectural Diagram

Figure 1: Network Architecture

3   Requirements and Specifications

3.1 General Requirements

[BBT/GR-001]   The device shall operate 24 hours a day, 7 days a week without the need to reboot.

[BBT/GR-002]   The device's power supply plug shall be male electrical connector of Type G (British 3-pin) BS 1363.

[BBT/GR-003]   The device's external power supply shall comply with the voltage and powering requirements of Cyprus Technical Norms and shall be certified to be used in the Cyprus electricity network.

[BBT/GR-004]   The device shall preserve local configuration information after a possible power-off or power interruption.

[BBT/GR-005]   The user shall be able to make calls by direct dialling the desired number typed in E.164 format.

[BBT/GR-006]   The user shall be able to read the local IP address of the phone in an easy way through the menu.

[BBT/GR-007]   In case the terminal/client supports supplementary services locally on the device via the dialling of specific Feature Access Codes (FAC), the user shall be able to activate and deactivate the support of these features.

[BBT/GR-008]   VoIP related passwords shall not be displayed nor broadcasted in any way by the device. This does not apply to the transmission of encrypted passwords used for network or session authentication.

[BBT/GR-009]   The Integrated Access Device shall support the triple play services, i.e., fast Internet (DSL Access), TV (Cytavision service) and VoIP (BBT service).

3.2 Interface Requirements

[BBT/IR-001]   IPVideophones and IPphones shall have an Ethernet port for connecting to the xDSL modem or to the Integrated Access Device.

[BBT/IR-002]   IPVideophones and IPphones shall be able to be interconnected to the xDSL modem or to the Integrated Access Device in order to make calls. The device shall have a 10/100 BASE-T Ethernet port (IEEE Std 802.3; RJ-45) for a WAN uplink to be connected to xDSL modem or Integrated Access Device.

3.3 Network Standards

[BBT/NS-001]   The terminal/client shall support the TCP, IP, UDP, routing and associated protocols identified here:

- IETF RFC 0768, User Datagram Protocol
- IETF RFC 0791, Internet Protocol
- IETF RFC 0792, Internet Control Message Protocol
- IETF RFC 0793, Transmission Control Protocol
- IETF RFC 0826, An Ethernet Address Resolution Protocol
- IETF RFC 0894, A Standard for the Transmission of IP Datagrams over Ethernet Networks
- IETF RFC 1042, A Standard for the Transmission of IP Datagrams over IEEE 802 Networks.

[BBT/NS-002]   The terminal/client shall support IP Version 4 (IETF RFC 791).

[BBT/NS-003]   The terminal/client shall support a DHCP client. This includes support for the following standards:

- IETF RFC 2131 Dynamic Host Configuration Protocol
- IETF RFC 2132 DHCP Options and BOOTP Vendor Extensions

[BBT/NS-004]   The Integrated Access Device shall be able to obtain IP network information dynamically on its connection to the broadband interface using DHCP. This information includes an IP address, primary and secondary DNS addresses, and a default gateway address. The terminal/client shall be able to obtain IP network information dynamically from the Integrated Access Device.

[BBT/NS-005]   The terminal/client shall repeat the DHCP requests in case of DHCP no reply.

3.4 Registration & Deregistration

[BBT/RD-001]   When configured, the terminal/client shall be able to REGISTER on the Registrar Server of the Ericsson IMS, IMT 3.0 network.

[BBT/RD-002]   In case of power-off and/or power interruptions, a SIP DEREGISTER and then DHCP Release message (RFC 2131) shall be sent.

[BBT/RD-003]   The client/terminal shall support the 503 Service Unavailable message and shall act according to the value included in the Retry-After header, according to RFC 3261.

3.5 Codec Requirements

[BBT/CR-001]   The terminal/client shall support ITU-T G.711 codec with 10, 20, and 30 ms frame sizes.

[BBT/CR-002]   The terminal/client shall support ITU-T G.711 Appendix I and ITU-T G.711 Appendix II.

[BBT/CR-003]   The terminal/client shall support A-law PCM encoding for G.711.

[BBT/CR-004]   The terminal/client shall support ITU-T G.729 Annex A with 10, 20, and 30 ms frame sizes.

[BBT/CR-005]   The terminal/client shall support configuration of a default codec to be used. The default codec is the first codec that is used during codec negotiation and selection.

[BBT/CR-006]   It shall be possible to configure the order of the codecs to be negotiated.

[BBT/CR-007]   When negotiating codec selection, the terminal/client shall use the first matching codec listed by the far end if the requested codec is available.

[BBT/CR-008]   The terminal/client shall support voice activity detection (VAD) on all connections using codecs supporting silence suppression, i.e., when using the G.729 codec that supports silence suppression, it implicitly applies that VAD shall be supported.

[BBT/CR-009]   The terminal/client shall be able to enable and disable silence suppression.

[BBT/CR-010]   The terminal/client shall support ITU-T G.168-compliant echo cancellation, including configurable tail-end delay up to at least 32 ms.

[BBT/CR-011]   The terminal/client shall support comfort noise generation.

[BBT/CR-012]   The terminal/client shall support a dynamic jitter buffer, which adjusts the size of the buffer based on detected delay.

[BBT/TS-013]   All the active calls shall be able to use the same codec individually.

[BBT/TS-014]   The IPVideophones shall support H.263 video codec. The IPVideophones shall support fallback to audio call mechanism which shall be activated in case the called party does not have video capabilities.

[BBT/TS-015]   The IPVideophones shall support H.264 video codec. The IPVideophones shall support fallback to audio call mechanism which shall be activated in case the called party does not have video capabilities.

[BBT/TS-016]   The IPVideophones shall support a number of Video bit rates such as 128, 256, 384, 512 kbps.

[BBT/TS-017]   The IPVideophones shall support a number of Video frame rates such as 10, 15, 20, 25, 30 frames/second.

3.6 Protocol Requirements

[BBT/PR-001]   The terminal/client must comply with RFC 3261; Session Initiation Protocol (SIP).

[BBT/PR-002]   The terminal/client must comply with RFC 3264: An offer/answer model with SDP.

[BBT/PR-003]   When receiving 180 Ringing, 183 Session Progress or 200 with SDP, the terminal/client shall open the RTP flows.

[BBT/PR-004]   When receiving 180 Ringing/SDP and then receiving 183 Session Progress/SDP with different/updated SDP the terminal/client shall use the latest SDP, i.e., that of the 183 Session Progress.

[BBT/PR-005]   When early media changes or if media changes upon transition to a confirmed dialog (that is, upon answer) the client shall process successive 18x responses where the To-tag and SDP changes. The client shall accept SDP changes upon call answer (for example, redirection to Voice Portal) in the 2xx response with a different To-tag.

[BBT/PR-006]   When receiving 180 without SDP, the terminal/client shall provide local ringing tone.

[BBT/PR-007]   The terminal/client shall support DTMF signalling according to RFC 2833; RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals.

[BBT/PR-008]   In addition to RFC 3261, the terminal/client must support authentication according to RFC 2617, HTTP Authentication; Basic and Digest Access Authentication. It should be noted, however, that it is only required to support the Digest Access Authentication Scheme.

[BBT/PR-009]   The terminal/client shall support sessions timers according to RFC 4028, Session Timers in the Session Initiation Protocol (SIP).

[BBT/PR-010]   The terminal/client must comply with RFC 3263; Locating SIP Servers.

| [BBT/PR-011] | The terminal/client shall provide support for Reliability of Provisional Responses in the Session Initiation Protocol (SIP) according to RFC 3262. The reliable responses are enabled for all calls, regardless of the transport type: udp or tcp. |

| [BBT/PR-012] | Support shall be provided for UPDATE method according to RFC 3311. The terminal/client shall support UPDATE, RFC3311 for early and established dialogues. If the terminal/client in an early dialogue uses UPDATE and for some reason (UPDATE not supported by terminating end-point) and receives 504, it shall not tear down the call. |

| [BBT/PR-013] | In case located behind a NAT router, the terminal/client should start sending "dummy RTP" data when receiving an "Early Media" answer from SDP, i.e. SDP in provisional response. The dummy data should be sent from the UDP-port where the client expects to receive "early data". |

| [BBT/PR-014] | The terminal/client MUST support Digest Authentication according to RFC 3261. |

| [BBT/PR-015] | The terminal/client should support user name in authentication headers in the form of "user@domain" where both the user portion and the host portion is present, according to the 3GPP specification of the Private User ID - 3GPP TS 23.003. It is recommended that the terminal/client has separate entries for user and domain portion of the private identity. |

| [BBT/PR-016] | The terminal/client shall support modification of existing dialogs using RE-INVITE as specified in RFC 3261. If the other party does not accept the change and sends a 488 (Not Acceptable Here) response, the client shall send and ACK and continue using the previously negotiated characteristics. Background: When *Emergency Call- deactivation on hold* is activated, the server will send 488 when the client /terminal tries to invoke HOLD. |

| [BBT/PR-017] | The terminal/client should support Warning header specified in RFC 3261. |

| [BBT/PR-018] | The Integrated Access Device shall be able to fallback to G.711 A-law for the transmission of fax messages. |

| [BBT/PR-019] | The terminal/client shall be able to send empty UDP packets if operating behind SIP/UDP, or newline characters if operating behind SIP/TCP, generally referred to as keep alive messages, in order to keep the NAT pinhole open when traffic is not sent. |

3.7 Supplementary Services

| [BBT/SS-001] | Hook Flash: The terminal/client shall send re-INVITE with HOLD SDP to the IMS network. The IMS network will respond with 200 OK. |

| [BBT/SS-002] | Call Waiting: IMS sends the Alert-Info header for any call which triggers distinctive ringing, independent of whether the phone is already involved |

in a call. This applies to FXS subscribers configured on Integrated Access Devices.

In all other cases (SIP terminal/clients behind IADs) the IMS will send an INVITE with OFFER SDP for the second call to the end-user terminal/client. The end-user terminal/client should send a re-INVITE with HOLD SDP to the IMS for the first call and a 200 OK with ANSWER SDP for the second call.

[BBT/SS-003]   Call Hold: The terminal/client device should send re-INVITE with HOLD SDP to the IMS. The IMS will respond with 200 OK.

[BBT/SS-004]   Three way calling: The terminal/client shall control the transfer and shall mix the media for the three-way call.