Product Specification

FRITZ!Box Fon 5113

Documentation release: April 22, 2009

Contents

1 System Components

1.1 Equipment

1.1-01	DSL modem router	
1.1-02	PBX for PSTN and Internet Telephony	
1.1-03	Network processor for bridging and routing	
1.1-04	SIP user agent client / SIP user agent server (Internet telephony)	

1.2 Ports

1.2-01	ADSL / ADSL 2+ Annex A	
1.2-02	PSTN subscriber line (ISDN and POTS)	
1.2-03	2 a/b lines (FXS) for analog telephone terminal devices (RJ11)	
1.2-04	1 Ethernet port	

1.3 Optional Variants

1.3-01	Product version for use with and without a fixed line, with different package contents (extra cables) and instructions	

1.4 Package Contents

1.4-01	FRITZ!Box Fon 5113	
1.4-02	DSL/TEL connecting cable (Y cable for fixed line and DSL)	
1.4-03	Analog line cable adapter (TAE plug/RJ45 socket)	
1.4-04	LAN cable	
1.4-05	Power supply plug	
1.4-06	Printed service information	

1.5 System Requirements

1.5-01	Internet line: Dual mode: ADSL or ADSL2+ line in accordance with Annex A or Ethernet connection 10/100 Base-T	
1.5-02	Telephony: - Internet telephony provider that supports Session Initiation Protocol (SIP) - analog telephone terminal devices	
1.5-03	On the LAN port (Ethernet, 10/100 base T): computer, notebook, Apple Macintosh, game console or other network devices	

2 Overview of Functions

2.1 ADSL2+ Modem

2.1-01	ADSL2+ compatibility: ITU G.992.1 Annex A (G.dmt), ITU G.994.1 (G.hs), ITU G.992.3 Annex A (ADSL2), ITU G.992.5 Annex A (ADSL2+)	
2.1-02	ATM compatibility: ATM-Forum UNI 3.1/4.0, UBR, OAM F4/F5 end-to-end/Segment loopback cells, AAL5, 8-bit VPI address range, VCI 16-bit address range	
2.1-03	Network compatibility: - ATM AAL5, PPPoE/PPPoA, PPP, IP - alternative: PPPoA, LLC, VC-Mux - alternative RFC 1483: bridged, routed IP LLC NLPID, routed IP LLC SNAP, routed IP raw	
2.1-04	RJ-45 socket, labeled DSL/TEL	
2.1-05	ADSL2+ monitoring: ADSL2+ line capacity, interleaved/fast-path, damping and SNR margin, ATM parameters including VPI and VCI, meter for different ATM cell types, for the transmission of used frequency ranges, etc.	
2.1-06	Layer 1 (ADSL DMT) can be updated for future demands	
2.1-07	Support for multiple ATM PVCs, for instance, for VoIP transmission	
2.1-08	VPI/VCI auto-detection	

2.2 DSL Router / DSL Bridge

2.2-01	IP packets from the LAN are routed through the PBX to the destination host	
2.2-02	PBX terminates an ADSL-PPPoE session	
2.2-03	DNS proxy/relay: local DNS (local IP address) with requests forwarded to the DNS conveyed by the Internet Service Provider via PPP	
2.2-04	DHCP server: automatic IP configuration of the computers/network devices connected via the LAN ports	
2.2-05	NTP client: NTP used for time synchronization (because of time stamp of logging)	
2.2-06	IP masquerading/NAT: - allows Internet sharing so that multiple computers can use just one IP address - port forwarding (reverse NAT) provides for safe and direct accessibility from the Internet of services on the local server - VPN pass through (IPSec, PPTP, L2TP)	
2.2-07	Protocols: - ARP - IP, ICMP, UDP, TCP, HTTP, DHCP, DNS relay, FTP	
2.2-08	MAC bridge LAN/ADSL2+: - Ethernet frames from the user interfaces are broadcast on ADSL/ADSL2+ (self-learning bridge, PPPoE filter) - application: Windows XP broadband connection, game console, etc. - multiple PPPoE sessions possible with one connection	
2.2-09	Universal Plug and Play (UPnP): - control of the shared use of the Internet connection by UPnP-capable devices, of port forwarding, for example - Internet Gateway Device (IGD) standardized device control protocol - overview of port forwarding settings configured via UPnP - multiple-step UPnP configuration (monitoring/control)	
2.2-10	Support for Multicast: - IGMP v1, v2, v3 - IGMP proxy - IGMP snooping - VLAN tagging	
2.2-11	Dynamic DNS: - automatic registration at DynDNS/Static IP (accessibility) - Dynamic DNS provider: dyndns.org, No-IP.com, DNS4BIZ.de, user-defined, etc.	

2.3 DSL Connection Manager

2.3-01	PPPoE client for the ADSL/ADSL2+ access network, (RFC 2516)	
2.3-02	PPP client for the ADSL/ADSL2+ Access Network (RFC 1618), PAP (RFC 1334), CHAP (RFC 1994)	
2.3-03	TCP/IP bandwidth management (DSL traffic shaping)	
2.3-04	Control of the PPP connection: - accessibility supported by holding the connection (always-on/auto-reconnect) - Short-hold mode (idle timer that adheres to rate interval) for time-based rates	
2.3-05	PPP monitoring/trace function (Etherreal)	
2.3-06	24-hour disconnect timeout can be delayed to a convenient time	
2.3-07	Support for multiple ATM PVCs, several PPPoE sessions and VLAN	

2.4 Firewall

2.4-01	Reliable Stateful Packet Inspection firewall	
2.4-02	IP masquerading - Network Address Translation with dynamic port assignment - local IP addresses/source ports remain invisible from outside - VPN pass through (IPSec, PPTP, L2TP)	
2.4-03	Stateful Packet Inspection Firewall: - only answers to LAN-side Internet requests are forwarded to LAN - port forwarding (incoming connections): selected ports for incoming data packets remain only open for the actual course of the connection (establishment, transmission, clearing) - Error Message Rate Limiter - port forwarding initialized with local IP address - port ranges can be enabled for forwarding - all ports enabled for forwarding on “exposed host”	
2.4-04	Incoming and outgoing packet filters: - NetBIOS over IP (Windows network operating system) - prevention of Denial of Service attacks - filters against incorrect arrivals of Peer-to-Peer traffic	

2.5 Local Area Network (Ethernet)

2.5-01	1 Ethernet port	
2.5-02	Connected computers/devices are networked with each other	
2.5-03	LAN interface: - 10/100 Mb/s IEEE 802.3/IEEE 802.3u (autosensing, Auto-MDIX) - port status	

2.6 PBX for Internet and PSTN Telephony

2.7-01	Internet telephony (Voice over IP)	
2.7-02	Connection to ISDN, analog fixed-line network/POTS and ADSL/ADSL2+	
2.7-03	POTS controller in accordance with Deutsche Telekom 1TR110 (May 2000)	
2.7-04	a/b interface (Fon 1 and Fon 2,) compliant with 1TR110	
2.7-05	PBX features available on both a/b ports	

2.6.1 Internet Telephony (Voice over IP)

2.7.1-01	Telephone calls via the Internet with connected analog telephones (local analog telephone interfaces, 2 a/b)	
2.7.1-02	Management of up to 10 SIP accounts (SIP addresses)	
2.7.1-03	All PBX system functions can be used (see Overview 2.7.4)	
2.7.1-04	SIP address assignment to the extensions	
2.7.1-05	Codecs: G.711a/µ, G.726-40, G.726-32, G.726-24, iLBC	
2.7.1-06	Voice transmission functions - Voice Activity Detection / Silence Suppression - Comfort Noise Generation (CNG) - Packet Loss Concealment (PLC) - Dynamic Jitter Buffer - Echo Cancellation (G.165, G.168) - DTMF – Inband / Outband (RFC 2833)	
2.7.1-07	SIP user agent client / SIP user agent server (Voice over IP) conform to the RFC 3261 standard: - RFC 3261 SIP: Session Initiation Protocol (SIP compliance in accordance with ETS 102 027-1 (2003-10) and IETF RFC 3261), see also Appendix C - RFC 2327 SDP: Session Description Protocol - RFC 2617: HTTP Authentication (Basic and Digest Access Authentication) - RFC 2782: A DNS RR for specifying the location of services (DNS SRV) - RFC 2976: The SIP INFO Method - RFC 3262: Reliability of Provisional Responses in Session Initiation Protocol (SIP) - RFC 3263: Session Initiation Protocol (SIP) - Locating SIP Servers - RFC 3264: An Offer/Answer Model with Session Description Protocol (SDP) - RFC 3265: Session Initiation Protocol (SIP) - Specific Event Notification - RFC 3311: The Session Initiation Protocol (SIP) UPDATE Method - RFC 3489: STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs) - RFC 3555: MIME Type Registration of RTP Payload Formats - RFC 3581: An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing - RFC 3665: Session Initiation Protocol (SIP) Basic Call Flow Examples - DRAFT draft-mahy-sip-signaled-digits	

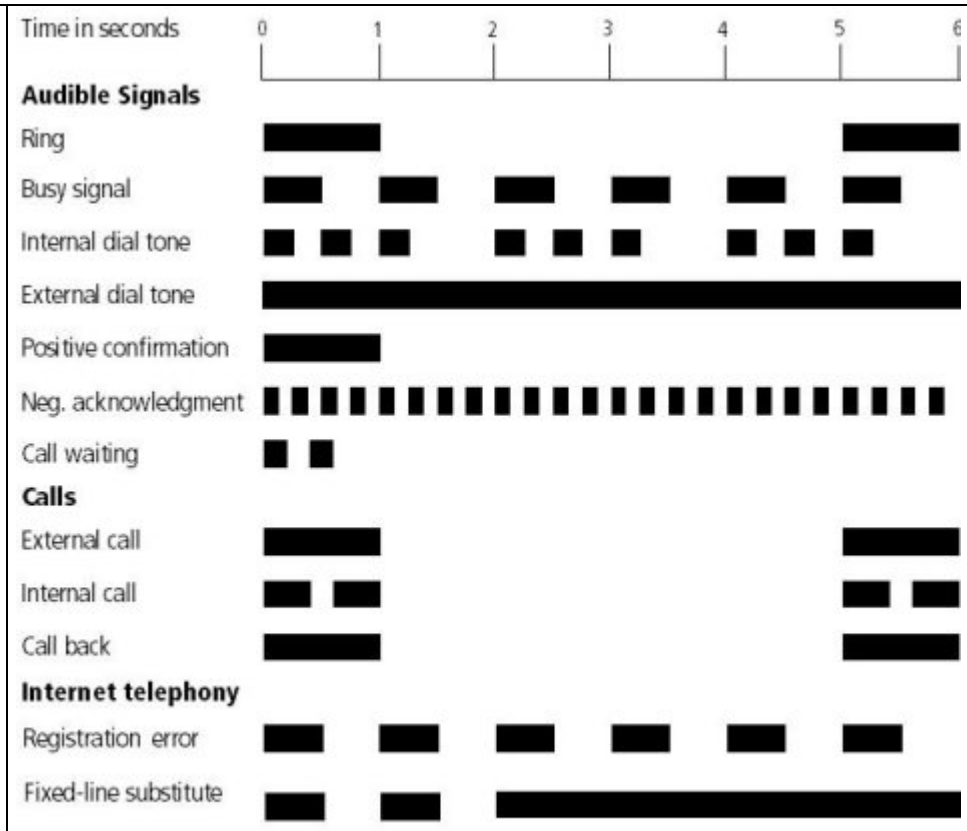
2.6.2 Features on the a/b Ports

2.7.2-01	a/b ports compliant with Deutsche Telekom 1TR110 (May 2000)	
2.7.2-02	Features on the network side can be used via analog, ISDN and VoIP (see Overview 2.7.4)	
2.7.2-03	Additional extension properties: - Do not disturb - parallel call - automatic outside dialing - baby monitoring phone function - alarm function - pickup - call waiting option - routing of calls to other extensions - extension naming	
2.7.2-04	Assignment of fixed-line and Internet numbers to the individual extensions: each extension can be assigned up to 10 MSNs	

2.6.3 PBX Features

2.7.3-01	Dialing rules: - outgoing number and type of connection for number ranges can be specified - number ranges can be defined for dialing over the PSTN line only; emergency numbers are preset - dialing using the fixed-line network or the Internet telephone numbers can be configured - carrier prefixes	
2.7.3-02	Manual selection of outgoing call number and connection type through the prefix	
2.7.3-03	Call routing / PSTN fallback: - if registration at the Internet telephony provider fails, the call will automatically be dialed using PSTN (configurable) - PSTN fallback for SIP Cause 403 (forbidden)	
2.7.3-04	Internal three-party conference: two parties can use the extensions to talk with an external party at the same time	
2.7.3-05	Call rejection when busy (busy on busy): this feature applies to the entire PBX: when it is enabled and one B channel is busy, a caller will always hear a busy signal	
2.7.3-06	Disable call waiting: while a call is in progress no external incoming calls will be signaled by call waiting; this feature can be enabled/disabled for each extension separately from the computer's web interface	

2.7.3-07	Picking up calls (pickup): a call at one extension can be picked up from another extension	
2.7.3-08	Quick-dial numbers: up to 40 quick-dial numbers and SIP addresses can be assigned for use at all extensions; quick-dial numbers are set up using a Web configuration assistant	
2.7.3-09	Call List: keeps track of incoming and outgoing telephone calls	
2.7.3-10	DTMF (Dual Tone Multi-Frequency): signal tones are generated by the telephones and sent to the remote site	
2.7.3-11	Blocking numbers: number ranges can be blocked for outgoing calls	
2.7.3-12	Click-to-dial: with the built-in dialer numbers from the Call List can be dialed with the click of a mouse	
2.7.3-13	Night service: switch off the ringer by enabling Do Not Disturb	
2.7.3-14	Telephone book: - storage of names and corresponding telephone numbers - assignment of a quick-dial number for each entry - name display of stored numbers during incoming and outgoing calls	
2.7.3-15	Treatment of callers: - rules tailored to individual callers - number-dependent rejection, diversion or patch-through despite Do Not Disturb	
2.7.3-16	Acoustic signaling: the diagram below shows the duration and the interval of the individual audio tones and call rhythms of a phone connected to an AVM FRITZ!Box:	



2.6.4 Supported Services and Features: Analog/ISDN/SIP

		analog	ISDN	SIP
2.7.4-01	MSN/SIP account (Multiple Subscriber Number)		•	•
2.7.4-02	Keypad	•	•	•
2.7.4-03	CLIP (Calling Line Identification Presentation: display of caller ID from party A to party B)	•	•	•
2.7.4-04	CLIR (Calling Line Identification Restriction: suppression of caller ID display from party A to party B)	•	•	•
2.7.4-05	HOLD (Hold/Consult/Transfer)	•	•	•
2.7.4-06	ECT (Explicit Call Transfer)	•	•	•
2.7.4-07	CW (Call Waiting)	•	•	•
2.7.4-08	3PTY (three-party conference)	•	•	•
2.7.4-09	TP (Terminal Portability: Suspend)		•	
2.7.4-10	MFV (DTMF signaling)	•	•	•
2.7.4-11	Pick up	•	•	•
2.7.4-12	Specified MSN assignment before dialing		•	•
2.7.4-13	Restore factory settings and delete all memory, test mode	•	•	•
2.7.4-14	Call List	•	•	•

2.7 Quality of Service (QoS)

2.8-01	Bandwidth manager for voice, TV and data via DSL/QoS (Quality of Service)	
2.8-02	VoIP prioritization / traffic shaping (bandwidth optimization)	
2.8-03	Download/upload can be limited during connection	
2.8-04	Automatic codec selection adapted to bandwidth	
2.8-05	Type of Service Support (ToS)/DiffServ	
2.8-06	Support for multiple ATM PVCs, VLAN tagging	

2.8 CPE WAN Management Protocol

2.09-01	CPE WAN Management Protocol, DSL Forum TR-069, May 2004, auto-configuration and dynamic service provisioning	
2.09-02	CPE methods supported in accordance with TR-069: GetRPCMethods, GetParameterNames, GetParameterValues, SetParameterValues, GetParameterAttributes, SetParameterAttributes, AddObject, DeleteObject, Download, FactoryReset, ScheduleInform	
2.09-03	Server methods supported in accordance with TR-069: Inform, TransferComplete	
2.09-04	Internet Gateway Device Version 1.1 Data Model (TR-098): profiles supported in accordance with TR-098: Baseline, EthernetLAN, ADSLWAN, IPPing, Wi-FiLAN	
2.09-05	DSL Home Provisioning Parameters for VoIP CPE, DSL-Forum TR-104, September 2005; extension of TR-069 regarding VoIP CPE; profiles supported in accordance with TR-104: Endpoint, SIP Endpoint	
2.09-06	Successful interoperability with leading ACS manufacturers (Plugfest UNH)	

3 Launching Operation and the User Interface

3.1 Assisted Launch

3.1-01	Help for starting operation: - clear Quick Guide - color-coded sockets and cables	
3.1-02	Configuration Wizard integrated in FRITZ!Box: - configuration of Internet access including connection check - manual configuration of VoIP connections including configuration check	
3.1-03	Automatic VoIP configuration: - automatic completion of VoIP number with country code and area code before dialing - switches between VoIP and fixed-line network if selected type of connection is not available (fallback) - dynamic dialing rules: call redirection to fixed-line network if subscriber cannot be reached via VoIP - automatic dial-in routing	
3.1-04	Automatic configuration and dynamic service provisioning on the basis of the CPE WAN Management protocol (DSL-Forum TR-069 / TR-104); see also 2.9	

3.2 Configuration

3.2.1 Via HTTP Server (HTML, JavaScript)

3.2.1-01	Internet: - account information (short-hold mode, automatic disconnection) - DSL settings for experts: VP (VCI, encapsulations) - Port forwarding - Dynamic DNS	
3.2.1-02	Telephony: - account information for registration with the Internet telephony provider - configuration of the fixed-line and Internet numbers - services can be configured individually (CLIP, call waiting, ...) - extension configuration - dialing rules - Telephone book - Call List (can be saved as an MS Excel list and used for dialing) - block against anonymous VoIP calls - Do Not Disturb - alarm - numbers routed to the fixed-line network (emergency calls, special numbers, international numbers) - settings can be configured on the computer; some directly using the telephone keypad	
3.2.1-04	System: - network devices (restriction of access rights) - network settings (UPnP, FRITZ!Box IP addresses) - FRITZ!Box password - firmware update (one-click option) - reset (restore factory settings) - save settings - Expert Mode (“normal” and “advanced” settings)	
3.2.1-05	Help	
3.2.1-06	configurable “INFO” LED	
3.2.1-07	Configuration Wizard: guided basic configuration including security	

3.2.2 Via UPnP (IGD)

3.2.2-01	Manual connection control	
3.2.2-02	Port forwarding	

3.2.3 Via Keypad

3.2.3-01	The complete list of the telephone keypad codes is recorded in the FRITZ!Box manual	

3.2.4 Via CPE WAN Management Protocol

3.2.4-01	Automatic configuration and dynamic service provisioning on the basis of the CPE WAN Management protocol (DSL-Forum TR-069 / TR-104); see also 2.9	

3.3 Monitoring

3.3.1 LED Display

3.3.1-01	Power/DSL, ‹Internet call›, ‹PSTN/analog line call›, Internet, INFO (configurable)	

3.3.2 Via Web Browser (HTTP)

3.3.2-01	Event Log: Internet connections (time, duration, volume), error conditions	
3.3.2-02	Call List	
3.3.2-03	Self-diagnosis (ADSL/ADSL2+, ATM, PPPoE, local account information, WebWatch)	
3.3.2-04	PPPoE traces (ethereal)	

3.4 Event Messaging

3.4-01	Events displayed optically with the "INFO" LED on the device	
3.4-02	Display messages	

4 Declaration of CE Conformity

4.1 Directives

4.1-01	1999/5/EEC R&TTE Directive: Radio Equipment and Telecommunications Terminal Equipment	
4.1-02	89/336/EEC EMC Directive: Electromagnetic Compatibility	
4.1-03	73/23/EEC Low Voltage Directive: Electric Equipment for Use	

4.2 Norms for Evaluation of Conformity

4.2-01	CTR 3/1998.06.17	
4.2-02	EN 55024/9.98 + A1/10.01+A2/01.03	
4.2-03	EN 301489-1 V1.5.1 (2004)	
4.2-04	EN 301489-17 V1.2.1 (2002)	
4.2-05	EN 60950-1:2001/A11	
4.2-06	ETSI TS 101 388, ITU-T G.992.1, ITU-T G.992.5, ITU-T G.994.1, ETSI ETR328	
4.2-07	EN 300328 V1.6.1 (11.2004)	

Appendix A: TÜV Test Report FRITZ!Box Firewall

Prüfergebnis

Sicherheitstechnischen Analyse der Firewall-Funktion der AVM FRITZ!Box

bei der

AVM Audiovisuelles Marketing und Computersysteme GmbH
Alt-Moabit 95
D-10559 Berlin

Die Firewall-Funktion der FRITZ!Box, wurde mit Methoden der Network Penetration Analysis in Bezug auf Schwachstellen untersucht, die zu einer Umgehung der eingestellten Sicherungsmechanismen führen könnten oder eine Gefährdung des geschützten Arbeitsplatzes darstellen. Die Funktion „FRITZ!webProtect" wurde nicht untersucht.

Unter anderem wurde der Schutz vor folgenden Angriffen aus dem Internet verifiziert:

- Schutz vor Internet-Würmern (z.B. W32.Blaster, ...), da die Ports 135, 139 und 445 zuverlässig geschlossen werden.
- Schutz vor Windows Messaging Pop-Ups, da Port 139 wirksam geschlossen wird.
- Schutz vor Fernsteuerung des Arbeitsplatzes, da alle Dienste wirksam geblockt werden können.
- Der Stateful Inspection Paketfilter sorgt dafür, dass keine nicht gewünschten Dienste des Arbeitsplatzes erreichbar sind.
- Schutz vor Denial of Service Angriffen, da die Anzahl der Antwortpakete limitiert wird und Pakete auf Standard-Konformität/-Konsistenz geprüft werden. (z.B. Ping-of-Death, Pingflooding, SYN-Flooding)

Die Firewall-Funktion der FRITZ!Box erwies sich robust gegenüber diesen und den im Bericht Nr. 63000712-01-04 der TÜV Secure iT GmbH beschriebenen Angriffen und gewährleistet im Regelbetrieb einen zuverlässigen Schutz gegenüber Angriffen, die eine eingehende Verbindung zum geschützten Arbeitsplatz hin erfordern.

Appendix B: SIP Conformity

Deutsche Telekom AG
T-Com Zentrale, Bereich TE4
Akkreditiertes Testlabor Nürnberg

CONFIRMATION OF TEST RESULTS

System under Test (SUT)

Name:	**FRITZ ! Box Fon**
Version:	**AVM SIP 3.01.03**
Supplier:	**Fa. AVM Audiovisuelles Marketing und Computersysteme GmbH**
Dates of Testing:	August / 31 / 2005 – September / 02 / 2005
Performed test services :	SIP Conformance Test according to ETS 102 027-1 V2.1.1 (2003-10) based on RFC 3261
Test Facilities Required:	Accredited Test Laboratory Nuremberg
Test Laboratory	
Testing laboratory:	Deutsche Telekom AG T-Com Zentrale, TE45 Akkreditiertes Testlabor Hansastraße 39 90441 Nürnberg
Accredited by:	DATech, represented in the:
Registration Number:	DAT-P-187/95-02
Statement:	The IUT has not been shown by conformance assessment to be non-conforming to the referenced protocol specification. *TCP is not supported as transport layer !*
Test Laboratory Manager:	Werner Mayer
Representative:	Jürgen Kupfer
Date:	September / 02 / 2005

Home address: Deutsche Telekom AG, T-Com Zentrale,
Department TE45, Hansastraße 39, 90441 Nuremberg
Postal : PO box 90 01 10, 90492 Nuremberg
Telecontacts: phone: +49 (0)911 6910 7416 +49 (0) 911 6910 7433 +49 (0) 911 6910 9270
e-mail: Werner.Mayer@t-com.net Juergen.Kupfer@t-com.net Gerald.Nun@t-com.net