

# Addpac

Настройка с помощью веб-интерфейса.

IP-адрес на LAN1 по умолчанию 192.168.10.1

Имя пользователя и пароль на всех устройствах Addpac: root / router

Заходим на веб интерфейс <http://192.168.10.1>

The screenshot displays the 'Smart Web Manager' web interface for an AddPac device. The main content area is titled 'System Information' and contains a table of system parameters. On the left, there is a navigation menu with categories: System, Basic, and Advanced. On the right, there are two side panels: 'Information' and 'Description'.

Parameter	Value
H/W Version	2.0
S/W Version	8.51.002
MAC Address	0002.a407.b3fe
VoIP Protocol	SIP
Voice Interface Module	G(2)O(2)
Registration Status	Unregistered
Supported Codec List	
Network Information	DHCP 192.168.1.209
WAN LINK Status	100Mbps FULL Duplex Link UP
LAN LINK Status	Link Down
Current Time	Fri Jan 14 18:10:40 2011
System Startup Time	Thu Jan 13 22:32:10 2011
System Running Time	0 days 19:38:30
Total Calls	0

**Information Panel:**  
AddPac Technology  
Model : GS1002\_G2  
H/W Version : 2.0  
S/W Version : 8.51.002  
Smart Web Version : 0.4  
Smart Web Build : Nov 23 2010  
Voice Interface  
G(2)O(2)  
Protocol : SIP  
Status : Unregistered  
Current Calls : 0  
CallNetwork : DHCP  
192.168.1.209  
Mac Address: 0002.a407.b3fe  
Unread Message:  
P0:0(3)  
P0:1(0)

**Description Panel:**  
Description

## Настраиваем SIP сервер

Use SIP Server = YES

Primary SIP Server = 192.168.1.254 (IP-адрес вашего SIP сервера или домен)

The screenshot shows the 'SIP (Session Initiation Protocol)' configuration page in the Smart Web Manager. The interface is divided into several sections:

- System:** Language, WAN Setup, LAN Setup, NAT, NTP, System Time, File Browser.
- Basic:** Protocol, **Server SIP**, Server H.323, SIP Registration, H.323 Registration, FXS Extension, FXO Extension, GSM Extension, DTMF/CODEC, VoIP Dial Plan, FXO DialPlan, GSM Dial Plan, Static Route, Hot Line.
- Advanced:** Gain & CID, GSM PINs, GSM USSD, Fax.

The main configuration area for SIP includes the following fields and options:

- Use SIP Server:** Radio buttons for Yes (selected) and No.
- Primary SIP Server:** Text input for IP address (192.168.1.254) and port (5060). Label: Server address and Port (default 5060).
- Secondary SIP Server:** Text input for IP address and port (5060). Label: Server address and Port (default 5060).
- Local Domain Name:** Text input for domain name. Label: (SIP userpart of authentication).
- SIP Signaling Port:** Text input for port (5060). Label: (default 5060, between 1 to 65535).
- Register Expiration:** Text input for seconds (60). Label: (in seconds, default 60, between 10 to 86400).
- Session Re-Fresh:** Radio buttons for INVITE (selected) and UPDATE.
- Session Expire Time:** Text input for seconds (1800). Label: (in seconds, default 1800, between 30 to 86400, 0 = disable).
- Min-SE:** Text input for seconds (1800). Label: (in seconds, default 1800, between 30 to 86400).

An 'Apply' button is located at the bottom of the configuration area.

**Information:**

- AddPac Technology
- Model : GS1002\_G2
- H/W Version : 2.0
- S/W Version : 8.51.002
- Smart Web Version : 0.4
- Smart Web Build : Nov 23 2010
- Voice Interface: G(2)Q(2)
- Protocol : SIP
- Status : Unregistered
- Current Calls : 0
- CallNetwork : DHCP
- 192.168.1.209
- Mac Address: 0002.a407.b3fe
- Unread Message: P0:0(1), P0:1(0)

**Description:** Configure the settings for SIP. Contact your service provider for the settings.

The bottom of the browser window shows a taskbar with several open files: asterikast-c...tar.gz, 59\_f9.pdf, PPTP-eng-v8.2....pdf, AddPac\_VoIP\_....pdf, and a 'Show all downloads...' button.

## Настраиваем GSM Extension

в разделе GSM Extension Configuration добавляем правила для исходящих вызовов. Выбираем каждый из портов и указываем Number = T

The screenshot displays the Smart Web Manager interface for configuring GSM Extension. The main content area is titled "GSM Extension" and contains several sections:

- Port Information:** A table showing port configurations for SLOT0.
- GSM Extension Configuration:** A table for configuring individual extensions.
- GSM Extension with Translation:** A table for defining translation rules.

On the left, a navigation menu includes sections for System, Basic, and Advanced settings. On the right, an "Information" panel provides system details, and a "Description" panel explains the configuration.

**Port Information Table:**

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXO	FXO

**GSM Extension Configuration Table:**

Pots Num	Port	Numbers	Preference	HuntStop	Control
3048	0/0	T	0	X	
3049	0/1	T	0	X	

**GSM Extension with Translation Table:**

Rule Num	Port	Destination Pattern	Digits to Insert	Digits to delete
900	P0:0	T		
901	P0:1			

Additional configuration elements include a "P0:0" dropdown menu, a "Delete" button, and an "Apply" button. Status messages indicate assigned tag numbers: "GSM Extension - Assigned Pots Tag Number : 3048 - 3559" and "GSM Extension - Assigned Translation-Rule Tag Number : 900 - 907".

# Настраиваем DTMF и кодеки для VoIP

Выбираем приоритет кодеков и DTMF режим.

The screenshot displays the Smart Web Manager interface for configuring DTMF and CODEC settings. The main content area is titled "DTMF/CODEC" and contains the following configuration options:

- Voice CODEC:** A list of six preferences, each with a dropdown menu:
  - Preference 1: g711alaw : G711 a-law Codec Type(64 kbps)
  - Preference 2: g711ulaw : G711 u-law Codec Type(64 kbps)
  - Preference 3: g7231r53 : G723.1 Codec Type(5.3 kbps)
  - Preference 4: g729 : G729 Codec Type(8 kbps)
  - Preference 5: g726r32 : G726 ADPCM Type(32 kbps)
  - Preference 6: None
- DTMF Relay mode:** Four radio button options:
  - DTMF relay by In-band voice
  - DTMF relay by RTP payload defined by RFC 2833
  - DTMF relay by Out-of-band signal
  - DTMF relay by Cisco out-of-band signal

Below the configuration options is a green "Apply" button with a checkmark icon. The left sidebar shows a navigation menu with categories: System, Basic, and Advanced. The right sidebar contains "Information" and "Description" sections. The bottom of the browser window shows a taskbar with several open files, including "asterikast-c...tar.gz", "59\_f9.pdf", "PPTP-eng-v8.2....pdf", and "AddPac\_VoIP\_....pdf".



Указываем Hotline номер (номер на который направить вызов сразу (без ожидания набора))  
SOP0(G) 8916..... (на этот номер будет послан вызов в VoIP сеть)

The screenshot displays the Smart Web Manager interface for configuring Hot Lines. The main content area is titled "Hot Line Configuration" and contains a table with the following structure:

Port	Hot Line Number	Digit Input Timeout <0-10 sec>
SOP0(G)	89167193410	0
SOP1(G)		0
SOP2(O)		0
SOP3(O)		0

Below the table is an "Apply" button with a green checkmark icon. The left sidebar contains navigation menus for "System", "Basic", and "Advanced" settings. The right sidebar shows "Information" and "Description" sections. The bottom of the interface features a taskbar with several open files: asterikast-c....tar.gz, 59\_f9.pdf, PPTP-eng-v8.2....pdf, and AddPac\_VoIP\_....pdf, along with a "Show all downloads..." link.