



# AP-3G1001™

## 1-Port 3G VoIP Gateway

High Performance 3G/UTMS VoIP Gateway Solution

Asterisk Free-PBX interworking 3G VoIP Gateway



**AddPac**

**AddPac Technology**

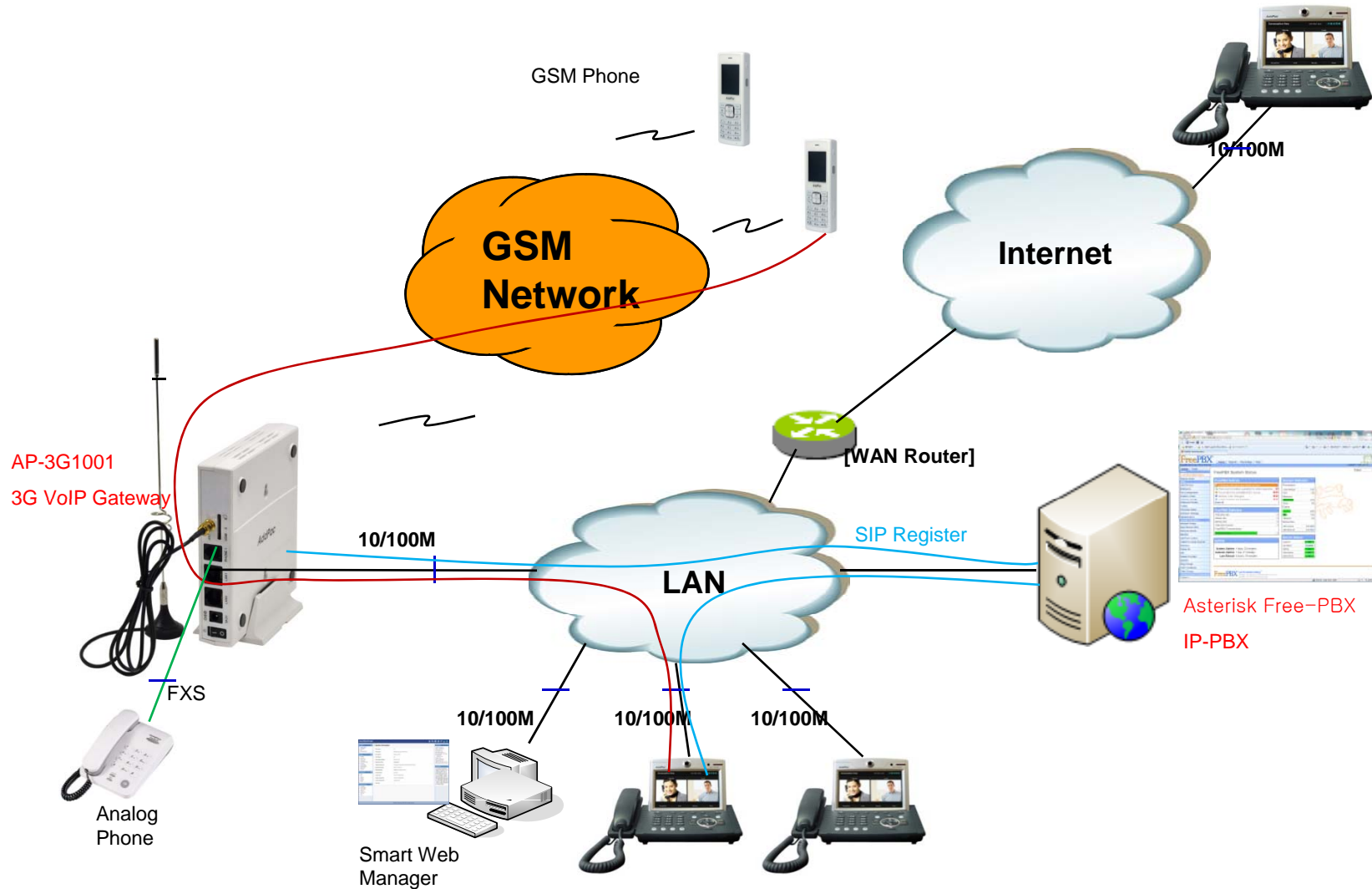
2012, Sales and Marketing

[www.addpac.com](http://www.addpac.com)

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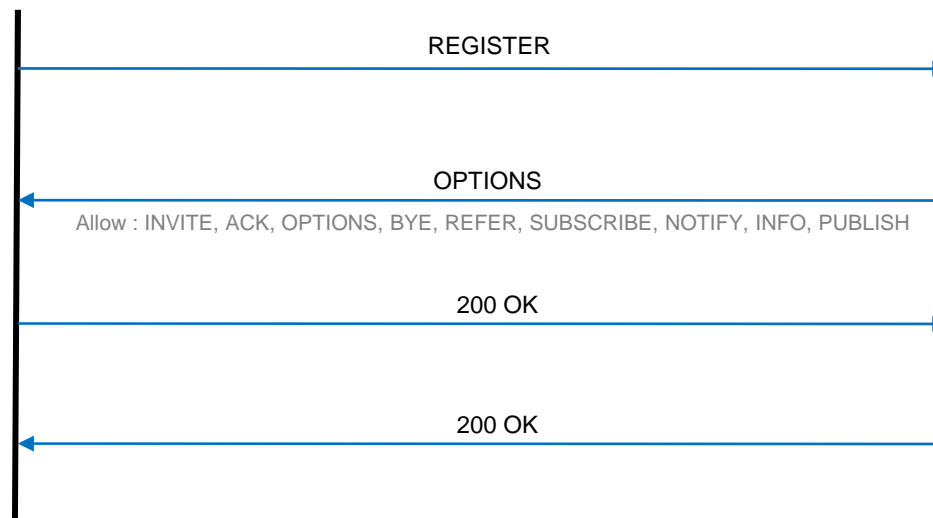
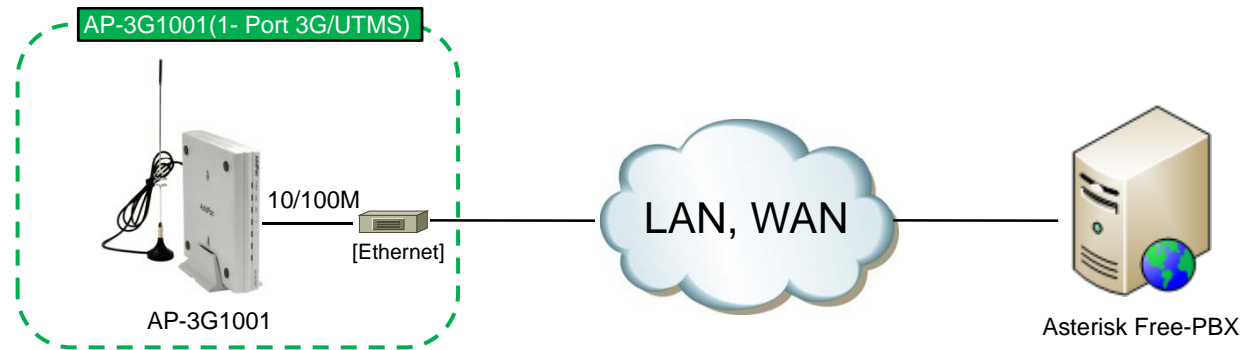
- Asterisk Free-PBX Interworking Network Diagram
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# Network Diagram



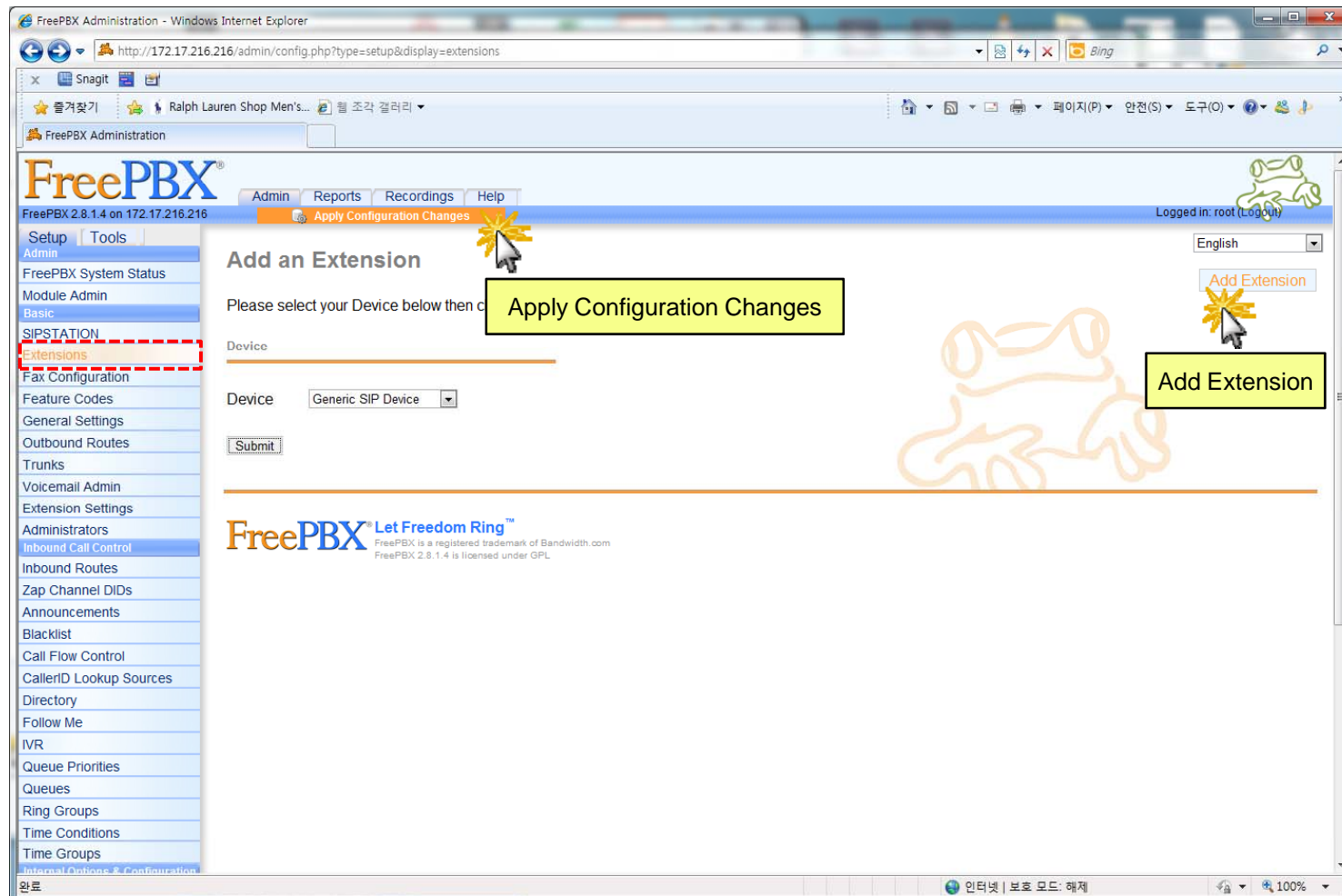
# SIP Register

## SIP Message Flow



# SIP Register

Asterisk Free-PBX Add Extension



# SIP Register

3G/UTMS Port Register to Asterisk Free-PBX

The screenshot shows the 'SIP (Session Initiation Protocol)' configuration page in the Smart Web Manager. The page is titled 'SIP (Session Initiation Protocol)' and contains several configuration fields. A red dashed box highlights the 'SIP Registration' section, which includes fields for 'Primary SIP Server', 'Secondary SIP Server', 'Local Domain Name', 'SIP Signaling Port', 'Register Expiration', 'Session Re-Fresh', and 'Session Expire Time'. A yellow callout box points to the 'SIP Registration' section in the left sidebar and contains the following text:

**SIP Server**  
Primary & Secondary server,  
Local domain name,  
SIP Signaling Port (reboot necessary)  
Timer  
\* register expire  
\* session refresh

Another yellow callout box points to the 'Primary SIP Server' field and contains the text:

Connect to GSM Gateway's IP Address

The right sidebar of the page contains 'Information' and 'Description' sections. The 'Information' section lists the following details:

- AddPac Technology
- Model: GS1002\_G2
- HW Version: 2.0
- SW Version: 0.51.008
- Smart Web Version: 0.4
- Smart Web Build: Apr 12 2012
- Voice Interface: G2/S2
- Protocol: SIP
- Status: Unregistered
- Current Calls: 0
- CallNetwork: DHCP
- 172.17.1.82
- Mac Address: 0002.a402.b97a
- Unread Message: P0:0)
- P0:10)

The 'Description' section contains the following text:

Configure the settings for SIP.  
Contact your service provider for the settings

# SIP Register

## 3G/UTMS Port Register to Asterisk Free-PBX

Connect to GSM Gateway's IP Address

Mobile Extension

Mobile Extension Configuration

Pots Num	Port	Numbers	Preference	HuntStop	Control
	P0:0	1000	0	<input type="checkbox"/>	Delete Apply

\* Mobile Extension - Assigned Pots Tag Number : 3048 - 3559

Mobile Extension with Translation

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

Sign-Rule Tag Number : 900 - 907

Information

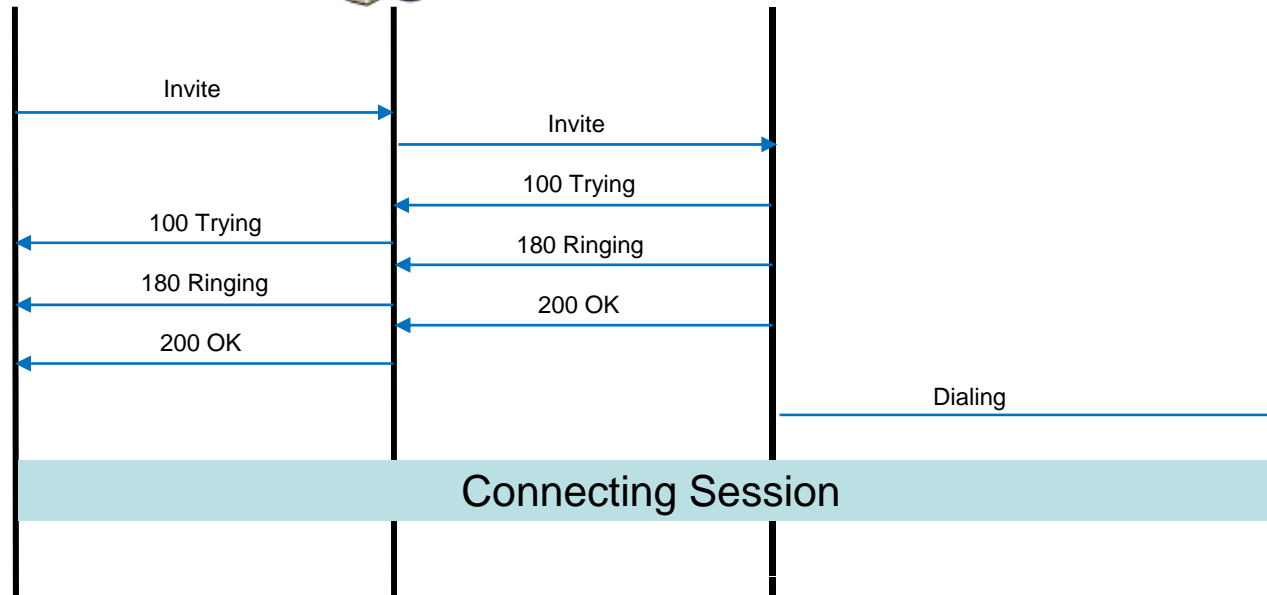
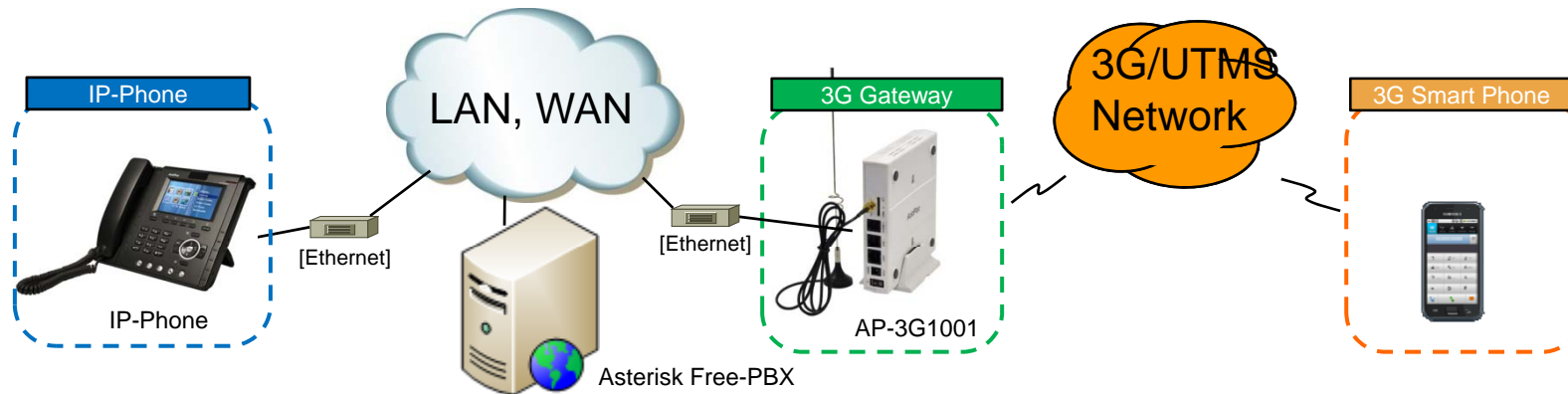
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172.17.1.82  
Mac Address : 0002.a402.b97a  
Unread Message : P0:0() P0:1()

Description

Set up for using Mobile port to extension number (forwarding No)

# Call Flow

## Asterisk Free-PBX Interworking 3G/UTMS VoIP Gateway







# Thank you!

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