

Model Name:	JIP4001/JIP4002	Object:	
Subject:	Release SIP Firmware of JIP4001/JIP4002 : sipfxso.103		
Release content			
Bug fixed			
<ol style="list-style-type: none"> 1. Fix some WEB display problems 2. The registration with Asterisk proxy. The registration with Asterisk proxy in previous version firmware will be failed caused by the Call ID issue. 3. PPPoE connection will be failed between JIP400x and some Linux PPPoE server. 4. If FXO greeting and then on-hook directly, it will cause JIP400x reboot automatically. 5. Fix the issue about 37 will not send unregistration under PPPoE mode. 6. Fix 37 fxs still hear ring back tone when he gets the 487 message. 7. The line2 ~line4 rtp port will error if open the ip-sharing function. 			
Featurea dn Function added			
<ol style="list-style-type: none"> 1. Add the "drop" function after the routing table identification. Command: route -add prefix 100 dst 1 e164 1001 min 3 max 7 hunt 0 drop 1 2. Add the "pause" command to delay send the DTMF through the FXO port during the one-stage-dialing. Command: pause -add prefix 9 delay 1 3. "Delay" function for FXO port. When FXO port has an incoming call from IP side and signal connection is established, it will wait the dial tone from PSTN or PBX. But sometimes the dial tone is too late and some errors will occur. Now user can use "sysconf -delay 2" command to change the time waiting for dial tone, where 2 is the delay time, default value is 1 second. 4. Auth command to let Root user customized the Administrator user authority. Command: auth -sip 1 5. Support the RFC2543 and RFC3261 transfers at the same time. 6. Add the 2nd proxy port function. Command: sip -px2 21.32.223.140 7. Add the 2nd proxy port change function. Command: sip -px2port 6000 8. Add the Outbound proxy port change function. Command: sip -outpxport 6000 			
Configuration item changed			
None			

Note

Modify the SIP local port range from 1024~65535 to 1~65535.