

Model Name:	JIP4001/JIP4002	Object:	*H.W. S.W.
Subject:	Release SIP Firmware of JIP4001/JIP4002 : sipfxso.104		
Release content			
Bug fixed			
<ol style="list-style-type: none"> 1. Fix the JIP400x shutdown issue when it is under Hotline mode. In the call load testing, the previous version will shutdown sometime. 2. The JIP400x uses In-band G711 for internal FAX. This feature is fail in sipfxo.103 and fixed in sipfxso.104. 3. The local SIP listen port is disappeared in p2p mode. 4. Fix the bug that disconnect tone frequency is not stable. But the ring cadence problem still exists. 5. Proxy may change the desination number in request line, but 37 will ignore it because it check the destination number is based on the to header. This version revises it. 6. Disable echo cancellation in greeting to avoid the “hang” problem and increase the stability of disconnect Tone analyzing. 7. Fix the secondary proxy problem. The from and to URL will still be the primary PX address when the proxy switched to 2nd px. 8. Fix the RFC2833 keypad, payload type 101 error in SDP. The payload type is still keep 96 in SDP of invite message although it is changed to 101. 9. Fix some DHCP error of vxworks. 			
Featurea dn Function added			
<ol style="list-style-type: none"> 1. Add the info dtmf. Command: sysconf –keypad 2 2. Add reverse function for FXS port 3. “FXO greeting” repeat 3 times and then return to IDLE when gets a PSTN incoming call. 4. Add the FXS DTMF callerID. 5. Add the CNG disable feature for G711 codec to fix the strange voice when 37 is using Asterisk proxy. Command: sysconf –silence 0 6. Add IVR disable function, using “dial tone” instead. Command: sysconf –ivr 0 			
Configuration item changed			
None			
Note			
<ol style="list-style-type: none"> 1. The routing table is still not base on “longest match” in this version. 2. Recording greeting still has some problems waiting for resolve. 			