Model Name:	JIP4001/JIP4002	Object:	H.W. *S.W.
Subject:	Release SIP Firmware of JIP4001/JIP4002 : sipfxso.102		

Release content

Bug fixed

1. **Registration.**
Add the MAX forward parameters in the registration request message.

2. **Hotline function.**
The FXO port will off hook the port if it got the connect message from the remote side

3. **Local no answer forward function.**
There will be the one-way audio problem if the no answer forward to the local port.

4. **Take off the relay changing function.**
Using the hotline function to replace the relay changing function if the registration wa failed.

5. **Ring detection for PSTN side.**
The accuracy for the ring detection isnot very good before. This version could detect th ring from the PSTN side correctly and filter the noise.

6. **Codec negotiation.**
FXSO will identify the SDP about the codec, if the codec couldn't be matched, it will rep the "busy here" message to the remote side.

7. **Dialing plan.**
The number will check out the prefix first and routing table later.

8. **Password configuration for the WEB.**
The password of the administrator is not working if userschange it. This was be fixed i this version.

9. **Flash clean command.**
This command will make all the configurations back to the default setting. This versio will keep the networking configuration.

Featurea dn Function added

1. Caller ID switches.

There is a switch to control the caller id sending function. It could support the FSK type as so far. (0 for disable ; 1 for FSK type)

2. Connect sending timer.

Under the one-stage-dialing function, the FXO port will detect the ring back tone and reversal signal to reply the calling party 200 message. Or users could define the timer for the 200 message sending. Even uses define the timer; the FXO port will still detect the ring back tone or reversal. (0 for detect the ring back tone or reversal signal only; 1~10 for sending 200 message timer.) – Still have the problems about the reversal signal detection.

3. SIP and Proxy port changing.

This version could change the SIP and Proxy port in this unit. For the Proxy port changing, the Proxy has to support this first.

4. Domain name for proxy address.

The proxy ip could be replaced by the domain name address.

5. Silence detection.

The FXO port could detect the silence from the PSTN side. It will drop the PSTN calls automatically if the FXO detect the silence for 40 seconds continuous. This function could be configured in the system configuration tables. This is the default enabled function without any command.

6. Call Forward and Transfer.

This version support Call forward (No answer forward, Busy forward and Unconditional forward) and call transfer function. Users could define the transfer type in the line configuration.

line –config 1 fwdtype 2 forward 123

fwdtype : 0 – Disable ; 1 – Unconditional ; 2 – Busy ; 3 – No answer

Unconditional : Every calls will be routed to the destination which you configured even that port is free.

Busy : The calls will be routed to the destination you set if the port is busy.

No answer : The calls will be routed to the destination which you set if the call didn't answer before the time is up. Users have to configure the timer for this function. Please configure the forward time in the system configuration.

About the no answer forward, the FXSO will check out the routing table first. If thereare no any rules for this call, the FXSO will send the 302 message to the proxy to make the calls reach another destination.

Configuration item changed
1. Changing the DNS address to 168.95.192.1 2. Changing the SNTP server address to 168.95.195.12 3. Changing the ring time in system configuration from 50 to 200 4. Changing the parameter of the ring before answer from 3 to 1 5. Changing the voice and input gain as following : FXO – 30, 32 6. Changing the Jitter buffer– min delay from 90ms to 60ms ; max delay from 150ms to 120ms. 7. Routing and prefix table support 30 records; Phone book support 40 records; FXO password support 5 records. 8. Changing the default SIP proxy ip to 10.1.1.2.
Note
1. Still have the problem about the reversal signal detection. 2. Couldn't support the Asterisk proxy because of the single Call-ID. 3. WEB Configurations will be change as same as the FXSO in H323 protocol. 4. The FXO port couldn't support the call transfer and forward function as so far.