

Configuration of Optimal Fax Completion Rates for SPA100 Series Adapters

Objective

Issues can occur with fax transmissions over IP networks. Adjustments can be made to several settings on the ATA (Analog Telephone Adapters) to optimize fax completion rates. These adjustments enhance the performance of fax transmission.

The objective of this document is to explain how to configure line settings to optimize fax completion rates on SPA100 Series Adapters.

Applicable Devices

- SPA112
- SPA122

Software Version

- 1.3.2-XU (014)

Optimal Fax Completion Rates

Step 1. Log in to the Phone Adapter Configuration Utility and choose **Voice > Line 1 or Line 2**. The *Line 1 or Line 2* page opens:

Line 1

General

Line Enable: yes

Streaming Audio Server (SAS)

SAS Enable: no

SAS Inbound RTP Sink:

SAS DLG Refresh Intvl: 30

NAT Settings

NAT Mapping Enable: no

NAT Keep Alive Msg: \$NOTIFY

NAT Keep Alive Enable: no

NAT Keep Alive Dest: \$PROXY

Network Settings

SIP ToS/DiffServ Value: 0x68

RTP ToS/DiffServ Value: 0xb8

Network Jitter Level: high

SIP CoS Value: 3 [0-7]

RTP CoS Value: 6 [0-7]

Jitter Buffer Adjustment: yes

SIP Settings

SIP Transport: UDP

SIP 100REL Enable: no

Auth Resync-Reboot: yes

SIP Remote-Party-ID: yes

SIP Debug Option: none

Restrict Source IP: no

Refer Target Bye Delay: 0

Refer-To Target Contact: no

Auth INVITE: no

Use Anonymous With RPID: yes

SIP Port: 5060

EXT SIP Port:

SIP Proxy-Require:

SIP GUID: no

RTP Log Intvl: 0

Referee Bye Delay: 4

Sticky 183: no

Reply 182 On Call Waiting: no

Use Local Addr In FROM: no

Call Feature Settings

Blind Attn-Xfer Enable: no

MOH Server:

Submit Cancel Refresh

Note: Make sure to choose the line for which adjustments are needed.

Step 2. Scroll down to the Network Settings section. Choose **Very High** from the Network

Jitter level drop-down list. Jitter is the variation in the time between packets arriving, caused by network congestion, timing drift, or route changes. Network Jitter determines how jitter buffer size is adjusted by the ATA.

NAT Keep Alive Msg:	SNOTIFY	NAT Keep Alive Dest:	SPROXY
Network Settings			
SIP ToS/DiffServ Value:	0x68	SIP CoS Value:	3 [0-7]
RTP ToS/DiffServ Value:	0xb8	RTP CoS Value:	6 [0-7]
Network Jitter Level:	very high	Jitter Buffer Adjustment:	no

Step 3. Choose **No** from the Jitter Buffer Adjustment drop-down list. This allows the Network Jitter Level to remain at its set value.

Supplementary Service Subscription	
Call Waiting Serv:	no
Block ANC Serv:	yes
Cfwrd All Serv:	yes
Cfwrd No Ans Serv:	yes
Cfwrd Last Serv:	yes
Accept Last Serv:	yes
CID Serv:	yes
Call Return Serv:	yes
Call Back Serv:	yes
Three Way Conf Serv:	no
Unattn Transfer Serv:	yes
VMWI Serv:	yes
Secure Call Serv:	yes
Feature Dial Serv:	yes
Reuse CID Number As Name:	yes
Block CID Serv:	yes
Dist Ring Serv:	yes
Cfwrd Busy Serv:	yes
Cfwrd Sel Serv:	yes
Block Last Serv:	yes
DND Serv:	yes
CWCID Serv:	yes
Call Redial Serv:	yes
Three Way Call Serv:	yes
Attn Transfer Serv:	yes
MWI Serv:	yes
Speed Dial Serv:	yes
Referral Serv:	yes
Service Announcement Serv:	no

Step 4. Choose **No** from the Call Waiting Serv drop-down list. This disables call waiting on the device.

Step 5. Choose **No** from the Three Way Call Serv drop-down list. This makes the user unable to have a conversation with two users at the same time.

Audio Configuration

Step 6. Scroll down to the Audio Configuration area. Choose either **G.711u** or **G.711a** from the Preferred Codec drop-down list. Codecs are protocols that allow the receiver to be able to reproduce the information exactly as it was sent. Both options are used for companding. In companding, the dynamic range of a signal is compressed before transmission and it is later expanded to reproduce the original information at the receiver.

Line 1

Audio Configuration	
Preferred Codec:	G711u
Third Preferred Codec:	Unspecified
Use Remote Pref Codec:	no
G729a Enable:	yes
G726-32 Enable:	yes
FAX V21 Detect Enable:	yes
FAX CNG Detect Enable:	yes
FAX Codec Symmetric:	yes
FAX Passthru Method:	ReINVITE
FAX Process NSE:	yes
FAX Disable ECAN:	no
DTMF Tx Strict Hold Off Time:	70
Hook Flash Tx Method:	None
FAX T38 ECM Enable:	yes
Symmetric RTP:	no
Modem Line:	no
Second Preferred Codec:	Unspecified
Use Pref Codec Only:	no
Codec Negotiation:	Default
Silence Supp Enable:	no
Silence Threshold:	medium
Echo Canc Enable:	no
FAX Passthru Codec:	G711u
DTMF Process INFO:	yes
DTMF Process AVT:	yes
DTMF Tx Method:	Auto
DTMF Tx Mode:	Strict
FAX Enable T38:	no
FAX T38 Redundancy:	1
FAX Tone Detect Mode:	caller or callee
FAX T38 Return to Voice:	no

- G.711u — -law encoding takes a 14-bit signed linear audio as input, increases the magnitude by 32, and converts it to an 8 bit value.
- G.711a — A-law encoding takes a 13-bit signed linear audio and converts it to an 8 bit value.

Step 7. Choose **Yes** from the Use Pref Codec Only drop-down list. This will ensure that all calls use only the preferred codec.

Step 8. Choose **No** from the Silence Supp Enable drop-down list. The silence suppression is used to avoid the transmission of silent audio frames over the network. This feature reduces the bandwidth of the network as only speech is transmitted.

Step 9. Choose **No** from the Echo Canc Enable drop-down list. The echo cancelling feature is used to remove the echo in the communication; this not only improves the quality of the call but also improves the silence suppression.

Step 10. Choose **ReINVITE** from the FAX Passthru Method drop-down list. This FAX pass through method is used to demodulate or compress the information that is passed through the network, the ReInvite method is used to send a message to the device so this sends an invitation to the host device to join to the network.

Step 11. Click **Submit** to save the settings or click **Cancel** to abandon the unsaved settings.