

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 ☒ no

Streaming Audio Server (SAS)

SAS Enable: ☐ no ☒ yes

SAS Inbound RTP Sink:

SAS DLG Refresh Intvl:

NAT Settings

NAT Mapping Enable: ☐ no ☒ yes

NAT Keep Alive Msg:

NAT Keep Alive Enable: ☐ no ☒ yes

NAT Keep Alive Dest:

Network Settings

SIP ToS/DiffServ Value:

RTP ToS/DiffServ Value:

Network Jitter Level:

SIP CoS Value: [0-7]

RTP CoS Value: [0-7]

Jitter Buffer Adjustment: ☐ yes ☒ no

SIP Settings

SIP Transport:

SIP 100REL Enable: ☐ no ☒ yes

Auth Resync-Reboot: ☐ yes ☒ no

SIP Remote-Party-ID: ☐ yes ☒ no

SIP Debug Option:

Restrict Source IP: ☐ no ☒ yes

Refer Target Bye Delay:

Refer-To Target Contact: ☐ no ☒ yes

Auth INVITE: ☐ no ☒ yes

Use Anonymous With RPID: ☐ yes ☒ no

SIP Port:

EXT SIP Port:

SIP Proxy-Require:

SIP GUID:

RTP Log Intvl:

Referor Bye Delay:

Referee Bye Delay:

Sticky 183: ☐ no ☒ yes

Reply 182 On Call Waiting: ☐ no ☒ yes

Use Local Addr In FROM: ☐ no ☒ yes

Call Feature Settings

Blind Attn-Xfer Enable: ☐ no ☒ yes

MOH Server:

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.

Line 1			
NAT Keep Alive Msg:	\$NOTIFY	NAT Keep Alive Dest:	\$PROXY
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.

Line 1	
Supplementary Service Subscription	
Call Waiting Serv:	no ▼
Block ANC Serv:	yes ▼
Cfwd All Serv:	yes ▼
Cfwd No Ans Serv:	yes ▼
Cfwd 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 ▼
Cfwd Busy Serv:	yes ▼
Cfwd 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.