

How to solve conference issues related to the RTP packetization interval

This article describes how issues related to the RTP Packetization interval can be solved in the case of conference calls on VoipNow servers. The recommended solution is to make sure that all parties involved in that call have the packetization interval set to **20ms** during conference calls on VoipNow servers. Otherwise, when channels are bridged, voice breaks, missing or dropped syllables, unexplained clicks or noises may occur.

Overview

Voice breaks, missing or dropped syllables, unexplained clicks or noises are triggered by different values of the `ptime` parameter when channels are bridged. According to the [RFC3551](#) for packetized audio, the default packetization interval must be 20 ms or one frame, whichever is longer, unless required otherwise. However, a receiver should be able to accept packages with a packetization interval between 0 and 200ms.

Good to know

If `autoframing=yes` is set in `sip.conf` within the global section, all calls will try to set the packetization based on the preferences of the remote end point.

When autoframing is enabled at global level, it applies to all regardless of the selected codec packetization.

If one of the parties involved in a call does not provide a `ptime` interval, the recommended value that should be used is **20ms**.

In VoipNow, autoframing is not enabled in `sip.conf`. Consequently, VoipNow will use default values for `ptime`, depending on the selected codec. For example, if a call is made using G711 as codec, the `ptime` will be 20ms. This means that each RTP packet sent during that call will only have **20ms** of audio recorded in it.

Ptime and different devices

Polycom IP 650/ Soundpoint 330,320

The `ptime` parameter for codecs can be set through provisioning by adding a specific value for each codec in the Device Configuration File #3:

```
voice.audioProfile.X.payloadSize="20"
```

The value for the payload size is in milliseconds: 10, 20, 30,...80;

X can be replaced by G711Mu, G711A, G722, G7221, and G729AB.

For example, when setting a `ptime=20` for the G711A codec, the following line should be added to the provisioning template:

```
voice.audioProfile.G711A.payloadSize="20"
```

Aastra 480i

- To set the `ptime` parameter on a Aastra 480i codec, you need to go to the **Advanced Settings->Global SIP->RTP Settings** area;
- Enter a **Customized Codec Preference List** which allows you to use the codecs that you prefer for this IP phone.

For example:

```
payload=8;ptime=20;  
payload=0;ptime=20;
```

Please note that `payload 0` is for G.711 u-Law and `payload 8` is for G.711 a-Law.

Status

System Information

OperationUser Password
Programmable Keys
Directory
Reset**Basic Settings**Preferences
Call Forward**Advanced Settings**Network
Global SIP
Line 1
Line 2
Line 3
Line 4
Line 5
Line 6
Line 7
Line 8
Line 9
Configuration Server
Firmware Update
Troubleshooting**Global SIP Settings****RTP Settings**RTP Port
Basic Codecs (G.711 u-Law, G.711 a-Law, G.729) Enabled
Force RFC2833 Out-of-Band DTMF Enabled
Customized Codec Preference List
DTMF Method **Aastra 6751i**

- To set the RTP packetization interval on Aastra 6751i, you need to go to the **Advanced Settings->Global SIP->Codec Preference List**;
- Valid values: 5 to 90, in increments of 5 milliseconds;
- To use this phone registered on a VoipNow server, you need to set the Packetization interval to **20ms**;

ASTRA 6751i
Log Off

Status
System Information

Operation
User Password
Phone Lock
Keypad Speed Dial
Directory
Reset

Basic Settings
Preferences
Account Configuration

Advanced Settings
Network
Global SIP
Action URI
Configuration Server
Firmware Update
TLS Support
802.1x Support
Troubleshooting

Global SIP Settings

RTP Settings

RTP Port: 3000

Force RFC2833 Out-of-Band DTMF: Enabled

DTMF Method: RTP

RTP Encryption: SRTP Disabled

Codec Preference List

Note: Basic Codecs Include G.711u (8K), G.711a (8K), G.729

Codec 1	G.722
Codec 2	None
Codec 3	None
Codec 4	None
Codec 5	None
Codec 6	None
Codec 7	None
Codec 8	None
Codec 9	None
Codec 10	None

Packetization Interval: 30

Silence Suppression: Enabled

Autodial Settings

Autodial Number:

Autodial Timeout:

Grandstream Gxp2020

Web Interface

Go to **Advanced Settings** -> **iLBC frame size** and select a framesize of 30ms.

Grandstream Device Configuration					
STATUS	BASIC SETTINGS	ADVANCED SETTINGS	EXT 1	EXT 2	
ACCOUNT 1	ACCOUNT 2	ACCOUNT 3	ACCOUNT 4	ACCOUNT 5	ACCOUNT 6
Admin Password: ●●●● (purposely not displayed for security protection)					
G723 rate: <input checked="" type="radio"/> 6.3kbps encoding rate <input type="radio"/> 5.3kbps encoding rate					
iLBC frame size: <input type="radio"/> 20ms <input checked="" type="radio"/> 30ms					
iLBC payload type: 97 (between 96 and 127, default is 97)					
Silence Suppression: <input checked="" type="radio"/> No <input type="radio"/> Yes					
Voice Frames per TX: 2 (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)					
Layer 3 QoS: 48 (Diff-Serv or Precedence value)					
Layer 2 QoS: 802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)					
Data VLAN Tag: 1: 0 2: 0 3: 0 (can't use the same non-zero value as 802.1Q tag)					
No Key Entry Timeout: 4 (in seconds, default is 4 seconds)					
Use # as Dial Key: <input type="radio"/> No <input checked="" type="radio"/> Yes					
local RTP port: 5004 (1024-65400, default 5004, must be even)					
Use random port: <input checked="" type="radio"/> No <input type="radio"/> Yes					
keep-alive interval: 20 (in seconds, default 20 seconds)					
Use NAT IP: (if specified, this will be used in SIP/SDP message)					
STUN server: stun.mycompany.com (URI or IP:port)					
Firmware Upgrade and Provisioning: Upgrade Via <input type="radio"/> TFTP <input checked="" type="radio"/> HTTP					
Firmware Server Path:					
Config Server Path:					
Firmware File Prefix:					
Firmware File Postfix:					
Config File Prefix:					
Config File Postfix:					
Allow DHCP Option43 and Option 66 to override server: <input checked="" type="radio"/> No <input type="radio"/> Yes					

Provisioning Template

You need to set the P97 variable to 1 as shown below:

```
# iLBC Frame Size. 0 - 20ms(default), 1 - 30ms.
P97 = 1
```

Yealink T20P

The ptime parameter can be set via provisioning templates.

In the device configuration file 1, you can set the ptime value as shown below:

```
[ account ]
path=/config/voip/sipAccount0.cfg
<yealink t20p configuration parameters>

ptime = 20

[ account ]
path=/config/voip/sipAccount1.cfg
<yealink t20p configuration parameters>

ptime = 20
```

Linksys PAP2

Some Linksys devices, such as PAP2 or SPA2102, have the RTP Packet Size parameter set to 30ms by default.

In such cases, the RTP Packet Size parameter can be changed from the **SIP** tab of the web interface.

For most users, the 0.030 factory default preset should be replaced with 0.020.

Please note that the RTP Packet Size parameter applies to all the lines served through that adapter.

LINKSYS
A Division of Cisco Systems, Inc. Firmware Version: 2.0.14(LS14)

Phone Adapter with 2 Ports for Voice-Over-IP PAP2

Voice

Info System **SIP** Provisioning Regional Line 1 Line 2 User 1 User 2

Advanced View [\(switch to basic view\)](#) User Login

SIP Parameters	SIP Timer Values (sec)	Response Status Code Handling	RTP Parameters	SDP Payload Types
Max Forward: <input type="text" value="70"/> Max Auth: <input type="text" value="2"/> SIP Server Name: <input type="text" value="\$VERSION"/> DTMF Relay MIME Type: <input type="text" value="application/dtmf-r"/> Remove Last Reg: <input type="text" value="no"/>	SIP T1: <input type="text" value="5"/> SIP T4: <input type="text" value="5"/> SIP Timer F: <input type="text" value="32"/> SIP Timer D: <input type="text" value="32"/> INVITE Expires: <input type="text" value="240"/> Reg Min Expires: <input type="text" value="1"/> Reg Retry Intvl: <input type="text" value="30"/>	SIT1 RSC: <input type="text"/> SIT3 RSC: <input type="text"/> Try Backup RSC: <input type="text"/>	RTP Port Min: <input type="text" value="16384"/> RTP Packet Size: <input type="text" value="0.03C"/> RTCP Tx Interval: <input type="text" value="0"/>	NSE Dynamic Payload: <input type="text" value="100"/> INFOREG Dynamic Payload: <input type="text"/>
Max Redirection: <input type="text" value="5"/> SIP User Agent Name: <input type="text" value="\$VERSION"/> SIP Accept Language: <input type="text"/> Hook Flash MIME Type: <input type="text" value="application/hook-"/> Use Compact Header: <input type="text" value="no"/>	SIP T2: <input type="text" value="4"/> SIP Timer B: <input type="text" value="32"/> SIP Timer H: <input type="text" value="32"/> SIP Timer J: <input type="text" value="32"/> ReINVITE Expires: <input type="text" value="30"/> Reg Max Expires: <input type="text" value="7200"/> Reg Retry Long Intvl: <input type="text" value="1200"/>	SIT2 RSC: <input type="text"/> SIT4 RSC: <input type="text"/> Retry Reg RSC: <input type="text"/>	RTP Port Max: <input type="text" value="16482"/> Max RTP ICMP Err: <input type="text" value="0"/>	AVT Dynamic Payload: <input type="text" value="101"/> G726r16 Dynamic Payload: <input type="text" value="98"/>

Siemens Speedstream 3610

To set the RTP packetization interval on Siemens Speedstream 3610, you need to go to the **Voice Setup -> Advanced Setting**.

If you are using this device registered on a VoipNow server, please set the RTP Packetization period to **20ms**. Valid values: in increments of 10ms.

SIEMENS Speedstream 3610 - Setup

Home | Data Setup | **Voice Setup** | Logout |

Advanced Setting

Configure the following VOIP-related parameters.

Support Call Waiting

Caller-ID Presentation

CID Type:

Support User-Agent Header

Support RFC 2833 (DTMF out of band)

FAX Passthrough Codec:

Conference Method:

Telephony Tone Country Setting:

Telephony Hook Flash Timer: ms

No Detection Less Than: ms

Dial Plan:

Replace '#' with ASCII code in Request - URI:

Voice Codec Configuration:

RTP Packetization Period: ms

Available Codecs: G.726-16, G.726-24, G.726-32, G.726-40

Selected Codecs: G.711 Mulaw, G.711 Alaw, G.729

Resources

Related articles

- [How to use the Conference features on a Phone terminal extension](#)
- [How to use the Conference feature in VoipNow](#)
- [How to solve audio issues caused by Asterisk timing configuration in VoipNow](#)
- [How to solve conference issues related to the RTP packetization interval](#)