



VoIP

VoIP	QoS
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SIP Account : SIP1

Active

SIP Settings

SIP Number	<input type="text" value="02888xxxx"/>
SIP Local Port	<input type="text" value="5060"/> <small>(1025-65535)</small>
SIP Server Address	<input type="text" value="85.119.188.3"/>
SIP Server Port	<input type="text" value="5060"/> <small>(1-65535)</small>
REGISTER Server Address	<input type="text" value="85.119.188.3"/>
REGISTER Server Port	<input type="text" value="5060"/> <small>(1-65535)</small>
SIP Service Domain	<input type="text" value="85.119.188.3"/>

Authentication

Authentication User-ID	<input type="text" value="02888xxxx"/>
Authentication Password	<input type="password"/>

Caller ID

Sending Caller ID

Incoming Call apply to

Phone1 Phone2

Advanced Settings





VoIP	QoS
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Advanced VoIP Settings :SIP1

SIP Server Settings

URL Type	SIP	
Expiration Duration	300	(20-65535)
Register Re-send timer	180	(1-65535)
Session Expires	180	(30-3600)
Min-SE	30	(20-1800)

RTP Port Range

From	50000	(1025-65535)
To	65535	(1025-65535)

Voice Compression

Preferred Compression Type	G.711 > G.729
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STUN

Active

Server Address	stun01.sipphone.com
Server Port	3478 (1024-65535)

Use NAT

Active

Server Address	
Server Port	5060 (1024-65535)

Outbound Proxy

Active

Server Address	
Server Port	3478 (1024-65535)

NAT Keep Alive

Enable NAT Keep Alive

Keep Alive Interval	120 (30-65535)
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Dual-Tone-Multi-Frequency (DTMF)

DTMF Mode


SIP INFO 

MWI (Message Waiting Indication) Enable**Expiration Time**

1800

(1-65535)

Fax Option Fax Pass-through T.38

Call Forward**Call Forward Table**Table 1 

Caller Ringing Enable**Caller Ringing Tone**Default 

On Hold Enable**On Hold Tone**Default **(Note: Currently works with calls that are using G.729 Codec)**