Recommended VTech ErisTerminal SIP phone configuration for Clearfly SIP service:

Note: *The following settings will establish basic two-way calling. The ErisTerminal phones are very flexible devices with many built-in capabilities, but advanced functionality was not part of this basic interop. Clearfly also provides two configuration templates at https://www.clearfly.net/support/interoperability/ (one each for 7- or 10-digit local dial plans) that can be uploaded via the manual provisioning section of the web interface and that will automatically configure all settings below.*

All configuration options were verified with firmware versions v1.1.4.x (VSP600, VSP715, VCS754) and v2.0.3.2 (VSP726, VSP736). If a setting is not displayed, default settings are acceptable.

SYSTEM->SIP Account Management->Account 1->General Account Settings

Setting	**Value**
Enable Account	*Checked*
Account Label	*<Often your phone number. Any text description works>*
User Identifier	*<Your phone number>*
Authentication Name	*<Your phone number>*
Authentication Password	*<Your Password (provided by Clearfly)>*
Dial Plan	*7-Digit:* *xx|[2-9]11|[2-9]xxxxxx|1[2-9]xx[2-9]xxxxxx|011x+ *10-Digit:* *xx|[2-9]11|[2-9]xx[2-9]xxxxxx|1[2-9]xx[2-9]xxxxxx|011x+
Inter-Digit Timeout (secs)	10 *(recommended)*
Maximum Number of Calls	2
Feature Synchronization	Disable
DTMF Method	RTP Event
Unregister After Reboot	Disable

SYSTEM->SIP Account Management->Account 1->SIP Server

Setting	**Value**
Server Address	v.cfly.co
Port	5060

SYSTEM->SIP Account Management->Account 1->Registration

Setting	**Value**
Server Address	v.cfly.co
Port	5060
Expiration (secs)	3600
Registration Freq (secs)	30

SYSTEM->SIP Account Management->Account 1->Outbound Proxy (and Backup Outbound Proxy)

Setting	**Value**
Server Address	<Leave Blank>
Port	5060

SYSTEM->SIP Account Management->Account 1->Caller Identity (v2.0.3.2)

Setting	Value
Source Priority 1	From
Source Priority 2	None
Source Priority 3	None

SYSTEM->SIP Account Management->Account 1->Audio

Setting	Value
Codec Priority 1	G.711u
Codec Priority 2	G.729a/b
Codec Priority [3-7]	None
Enable Voice Encryption (SRTP)	Unchecked
Enable G.729 Annex B	Unchecked
Preferred Packetization Time (ms)	20
DTMF Payload Type (v2.0.3.2)	101

SYSTEM->SIP Account Management->Account 1->Quality of Service

Setting	Value
DSCP (voice)	46
DSCP (signaling)	46

SYSTEM->SIP Account Management->Account 1->Signaling Settings

Setting	Value
Local SIP Port	5060
Transport	UDP

SYSTEM->SIP Account Management->Account 1->NAT Traversal

Setting	Value
Enable STUN	Unchecked
Enable UDP/STUN Keep-Alive	Unchecked

SYSTEM->SIP Account Management->Account 1->Voicemail Settings

Setting	Value
<All Settings>	<Leave as Default>

SYSTEM->SIP Account Management->Account 1->NAT Traversal

Setting	Value
<All Settings>	<Leave as Default>