Asterisk Business Edition™
Digium Partner Certification

Cyberdata VoIPSpeaker
Interoperability Report
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Executive Summary

Interoperability certification testing was conducted to ensure that Asterisk Business Edition® users will experience the level of quality and ease of use they expect from our business class IP PBX solution when integrated with CyberData VoIP Speaker.

This interoperability test report documents the detailed testing conducted to certify the Cyber Data VoIP Speaker for use with Asterisk Business Edition.

During testing by Digium® the most common features were exercised extensively in order to demonstrate full interoperability across supported features, the phone was configured as a typical user would require. Testing first covered SIP registration and common calling functions followed by more advanced feature testing. Further testing included long term call duration and audio quality measurements. This testing ensures features implemented in the CyberData VoIP Speaker functions as expected.

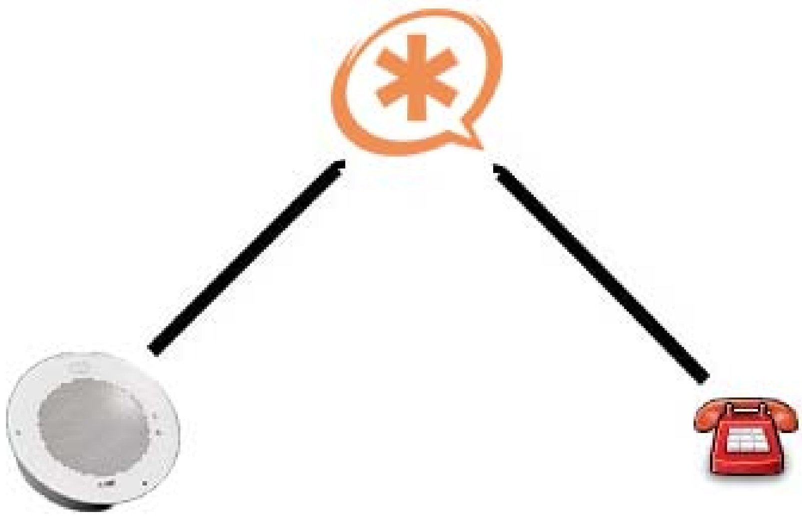

Figure 1: Test Configuration

Test Lab

The following functionality was tested:

Registration Test

	Test Step / Substep	Expected Action	Pass/Fail
1	Set SIP registration option	Verify speaker registered	Pass
2	Set SIP de-register option	Verify speaker de-registered on reboot.	Pass
3	Set registration expiration option to 1 minute	SIP phone re-registered once a minute.	Pass

Functional Call

	Test Step / Substep	Expected Action	Pass/Fail
1	Phone calls speaker	Beeps and answers	Pass
2	Playback calls speaker	Beeps then answers	Pass
3	Test turning beep off and on	Verify that speaker beeps at beginning of call only when option is set.	Pass

Functional Device Tests – Hardware Functionality

	Test Step / Substep	Expected Action	Pass/Fail
1	Holding RTFM switch for a minute resets firmware.	Verified default setup	Pass
2	Volume control works	Verified that knob did adjust volume	Pass
3	Admin password change	Change admin password	Pass

Glossary

Term	Definition
AP	Access Point. An 802.11 term for a station or transceiver (in a WLAN) that transmits and receives data among users in the network, and might also be the connection between the WLAN and a fixed-wire network. An AP serves multiple users within a portion of the network.
Codec	Coder/Decoder, Compressor/Decompressor. Software or hardware (or a combination of both) that converts data to a code and later decodes it, e.g. telephone firmware that converts digital signals to analog, and vice versa. Also, technology (such as MPEG) that compresses data (such as sound files) for storage and decompresses it for processing.
DND	Do Not Disturb
Enbloc Dialing	Method of initiating a call where the dialed digits are collected prior to call setup, e.g. dialing an extension then pressing the start key.
Fast Busy	A busy signal (also referred to as a “reorder”) in telephony is an audible or visual signal to the calling party that indicates failure to complete the requested connection of that particular telephone call.
Forked Call	When a proxy server sends an INVITE to a number of locations at the same time, this type of parallel search is known as forking.
Hang Up	Press the END key on the WT, or in some other way put a phone on-hook. (Compare to “Press END key”.)
LCD	Liquid Crystal Display; the 128-by-64-pixel display screen on SpectraLink Wts.
MAC	Media Access Control(ler)
MGCP	Media Gateway Control Protocol is a protocol used within a distributed Voice over IP system that can appear to the outside world as a single VoIP gateway.
On-hook, off-hook	Old telephony terms referring to the receiver being on the hook or cradle (inactive) and being off hook and in the user’s hand (actively engaged with the PBX). The terms have become nebulous in modern IP telephony, and their actual meaning depends largely on the nature of the attached “PBX,” which also has become a nebulous term (see below).
PBX	Private Branch Exchange. Originally referring to a system providing local telephone service (“public exchange”) and access to the PSTN, PBX now typically refers to whatever connection a phone user has to other users or to the outside world. In some cases, that connection is a call manager, call server, or gateway, or some other box or combination of boxes. In some IP protocols there might not even be such a box, but simply a direct access to the Internet.
PRI	Primary rate interface (PRI) is a telecommunications standard for carrying multiple DS0 voice and data transmissions between two physical locations.
Press END key	Press the END key on the WT. (Compare with “Send END_KEY to PBX”.)
PSTN	Public switched telephone network. The worldwide interconnection of all commercial and governmental public telephone networks. Sometimes called “POTS” (Plain Old Telephone Service).

Term	Definition
Reorder Tone	Reorder tone (also referred to as "fast busy") is a dual-frequency tone of 480 Hz and 620 Hz at a cadence of 0.25 seconds on, 0.25 off. It is used to indicate that a person has dialed an invalid code, or that all circuits (trunks) are busy and/or their call is unroutable.
SIP	Session Initiation Protocol (SIP) is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. SIP is an ASCII-based, application-layer control protocol (defined in RFC 2543) that can be used to establish, maintain, and terminate calls between two or more end points.
Softkey	Softkey is a button, located along the display device, which performs whatever function is shown near it on that display.
Standby	Standby pertains to a power-saving condition or status of operation of equipment that is ready for use but not in use.
TFTP	Trivial (or Thin) File Transport Protocol. A simple form of FTP, TFTP uses UDP and provides no security features. It is often used by servers to boot diskless workstations, X-terminals, and routers.
VoIP	Voice-over Internet Protocol

Appendix A – Configuration Files

Configuring CyberData VoIP Speaker for use with Asterisk

The CyberData VoIP Speaker is configured via a web interface. Boot the device and wait for the status led to turn green. Press the RTFM key, wait for the beep, and then release the key. The CyberData VoIP Speaker will then speak its IP address. From their you can go to http://IP and login default is (admin/admin).

sip server: asterisk

remote sip port: 5060

local sip port: 5060

sip user id: speaker

auth id: speaker

auth password: speaker

sip registeration: yes

unregister on boot: yes

Below are simplified configuration files for use with Asterisk Business Edition:

Extensions.conf

Edit the extensions.conf file. The following example instructs Asterisk to call the Cyberdat VoIP speaker.

Example:

exten => 1000,1,Dial(sip/speaker)

Sip.conf

Add the phone to the Sip.conf file

Example:

```
[speaker]
secret=speaker
host=dynamic
type=friend
disallow=all
allow=ulaw
```

Note: Restart Asterisk service to reload Asterisk process.