

NetLink



OFFICIAL SEGA SATURN *Online* GAMING OVER VOIP GUIDE

2.0

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REQUIRED HARDWARE

The hardware needed should be easy to find on **Ebay** or **Amazon**.

1. UNLOCKED Linksys SPA1001 FXS VoIP Phone Adapter
(try to find one that comes with a power adapter) - Other VoIP adapters may work, but they haven't been tested. As more are tested in the future they will be added to the list of confirmed working VoIP adapters.

2. Ethernet Cable (confirmed working)

3. RJ-11 Phone Cable (confirmed working)

* best to use one with only 2 contact pins at each end, meaning it only has 2 wires (send & receive) which may reduce line noise.

A. US NetLink Modem

Compatible with ALL PATCHED games (US & JP)

Compatible with ALL US RETAIL games, ***(performance will be reduced if using US NetLink modems on both ends of connection - See Page 12)**

B. JP XBand Modem

Compatible with ALL PATCHED games (US & JP)

Compatible with ALL US RETAIL games

*** STOCK JAPANESE RETAIL GAMES ARE NOT SUPPORTED!**

** for the purposes of this guide, "NetLink" refers to North American modem and/or games, and "XBand" refers to the Japanese modem and/or games.*

1



2



3



A OR B



OPTIONAL HARDWARE

For accessing NetLink Zone, Surfing the Internet, Sending Emails & Trading Files.

5. DreamPi Kit (Raspberry Pi & USB 56k Modem for connecting to online services)

6. RJ-11 Phone Line Switch (2 Port RJ11 RJ12 Manual Sharing Telephone Switch Box Phone selector) - Allows you to connect all equipment together, including Sega Dreamcast.

7. RJ-11 Phone Line Splitter (confirmed working)

5



6



7



HARDWARE SETUP (REQUIRED)

1. Run the Ethernet cable from your home network router into the Ethernet input on the VoIP adapter. *(make sure the VoIP adapter is powered)*
2. Plug one end of your telephone cord into the VoIP adapter and the other end into the "LINE" input on the Sega Saturn NetLink or JP XBand Modem.



HARDWARE SETUP (OPTIONAL)

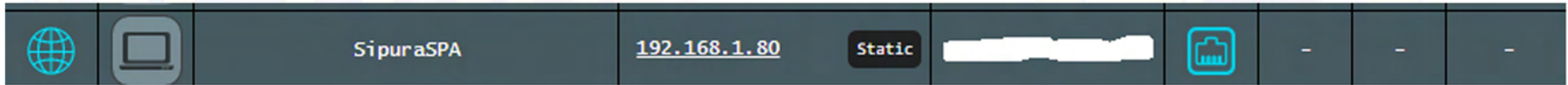
This section will show you how to setup your Sega Saturn to work with both a DreamPi and VoIP ATA Adapter together.

1. Plug Sega Saturn into socket 1 of the phone line splitter
2. (Optional) Plug your Sega Dreamcast into socket 2 of the phone line splitter
3. Plug your phone line splitter into IN/OUT port on Phone Line Switch
4. Plug DreamPi into Slot 1 on Phone Line Switch
5. Plug SPA1001 VoIP ATA into Slot 2 on Phone Line Switch
6. When you wish to connect to internet press button 1 on the switch, press 2 to use the VoIP ATA.



VOIP CONFIGURATION

1. You first need to login to your router and find out what IP Address your voip adapter was assigned.
2. In our case it was assigned 192.168.1.80, The device should also already be named "SipuraSPA" as seen in the screen shot below.



LOGIN

3. Type in that IP address into any browser and press enter. Then you'll need to click on "Admin Login" and then "Advanced" on the top right hand corner of your screen.



SYSTEM TAB

4. Click on System Tab (Optional Step)

Set DHCP to NO,
Setup a static IP of your choosing
Enter your routers Gateway
(Find this by using ipconfig on windows command prompt)
NetMask must be 255.255.255.0




SIP TAB

5. Click on the SIP Tab and Verify/Match Settings boxed in red in image below

Troubleshooting Scenario:

If your opponents netlink answers your call but never exchanges information, try setting "STUN Enable" to No and putting your public IP address in EXT IP



LINKSYS® A Division of Cisco Systems, Inc.		Linksys Phone Adapter Configuration	
Info	System	SIP	Provisioning Regional Phone Line 1 Line 2 User 1 User 2 Help Login Basic Advanced
SIP Parameters			
Max Forward:	70	Max Redirection:	5
Max Auth:	2	SIP User Agent Name:	\$VERSION
SIP Server Name:	\$VERSION	SIP Reg User Agent Name:	
SIP Accept Language:		DTMF Relay MIME Type:	application/dtmf-relay
Hook Flash MIME Type:	application/hook-flash	Remove Last Reg:	no ▼
Use Compact Header:	no ▼	Escape Display Name:	no ▼
RFC 2543 Call Hold:	yes ▼		
SIP Timer Values (sec)			
SIP T1:	.5	SIP T2:	4
SIP T4:	5	SIP Timer B:	32
SIP Timer F:	32	SIP Timer H:	32
SIP Timer D:	32	SIP Timer J:	32
INVITE Expires:	240	ReINVITE Expires:	30
Reg Min Expires:	1	Reg Max Expires:	7200
Reg Retry Intvl:	30	Reg Retry Long Intvl:	1200
Reg Retry Random Delay:		Reg Retry Long Random Delay:	
Reg Retry Intvl Cap:			
Response Status Code Handling			
SIT1 RSC:		SIT2 RSC:	
SIT3 RSC:		SIT4 RSC:	
Try Backup RSC:		Retry Reg RSC:	
RTP Parameters			
RTP Port Min:	16384	RTP Port Max:	16482
RTP Packet Size:	0.020	Max RTP ICMP Err:	0
RTCP Tx Interval:	0	No UDP Checksum:	no ▼
Stats In BYE:	no ▼		
SDP Payload Types			
NSE Dynamic Payload:	100	AVT Dynamic Payload:	101
INFOREQ Dynamic Payload:		G726r16 Dynamic Payload:	98
G726r24 Dynamic Payload:	97	G726r32 Dynamic Payload:	2
G726r40 Dynamic Payload:	96	G729b Dynamic Payload:	99
NSE Codec Name:	NSE	AVT Codec Name:	telephone-event
G711u Codec Name:	PCMU	G711a Codec Name:	PCMA
G726r16 Codec Name:	G726-16	G726r24 Codec Name:	G726-24
G726r32 Codec Name:	G726-32	G726r40 Codec Name:	G726-40
G729a Codec Name:	G729a	G729b Codec Name:	G729ab
G723 Codec Name:	G723		
NAT Support Parameters			
Handle VIA received:	yes ▼	Handle VIA rport:	yes ▼
Insert VIA received:	yes ▼	Insert VIA rport:	yes ▼
Substitute VIA Addr:	yes ▼	Send Resp To Src Port:	yes ▼
STUN Enable:	yes ▼	STUN Test Enable:	yes ▼
STUN Server:	stun.voiparound.com	EXT IP:	
EXT RTP Port Min:	16384	NAT Keep Alive Intvl:	15

PROVISIONING TAB

6. Click on Provisioning Tab and Verify/Match the Settings in the red box here:



LINKSYS®
A Division of Cisco Systems, Inc.

Linksys Phone Adapter Configuration

Info System SIP **Provisioning** Regional Phone Line 1 Line 2 User 1 User 2 [User Login](#) [Basic](#) | [Advanced](#)

Configuration Profile

Provision Enable:	no ▼	Resync On Reset:	yes ▼
Resync Random Delay:	2	Resync Periodic:	3600
Resync Error Retry Delay:	3600	Forced Resync Delay:	14400

REGIONAL TAB

7. Click on Regional Tab and copy the following line into "Reorder Tone" field:

480@-19,620@-19;10(.5/.5/1+2)



LINKSYS®
A Division of Cisco Systems, Inc.

Linksys Phone Adapter Configuration

Info System SIP Provisioning **Regional** Phone Line 1 Line 2 User 1 User 2 [User Login](#) [Basic](#) | [Advanced](#)

Call Progress Tones

Dial Tone:	350@-19,440@-19;10(*/*0/1+2)
Second Dial Tone:	420@-19,520@-19;10(*/*0/1+2)
Outside Dial Tone:	420@-16;10(*/*0/1)
Prompt Tone:	520@-19,620@-19;10(*/*0/1+2)
Busy Tone:	480@-19,620@-19;10(.5/.5/1+2)
Reorder Tone:	480@-19,620@-19;10(.5/.5/1+2)
Off Hook Warning Tone:	480@-10,620@0;10(.125/.125/1+2)
Ring Back Tone:	440@-19,480@-19;*/*7/1/1+2)

PHONE TAB

8. Click on Phone Tab and Verify/Match the Settings in the red box here:



LINKSYS®
A Division of Cisco Systems, Inc.

Linksys Phone Adapter Configuration

Info System SIP Provisioning Regional **Phone** Line 1 Line 2 User 1 User 2 [User Login](#) [Basic](#) | [Advanced](#)

Line Selection

Default Line:	Line 1 ▼	Line 1 Select Code:	
Alt Dial Tone Multiplier:	1.5	Line 2 Select Code:	#

Streaming Audio Server (SAS)

SAS Enable:	no ▼	SAS DLG Refresh Intvl:	30
SAS Inbound RTP Sink:			

Audio Configuration

Echo Canc Enable:	no ▼	FAX CED Detect Enable:	no ▼
Echo Canc Adapt Enable:	no ▼	FAX CNG Detect Enable:	no ▼
Echo Supp Enable:	no ▼	FAX Disable ECAN:	no ▼

FXS Port Polarity Configuration

Idle Polarity:	Forward ▼	Caller Conn Polarity:	Forward ▼
Callee Conn Polarity:	Forward ▼		

LINE 1 TAB

9. Click on Line 1 Tab and Verify/Match the Settings in the red boxes here:

Troubleshooting Scenarios:

If your opponents netlink never "Answers" when you dial or are being dialed then your ISP could be blocking port 5060. In this case you'll need to change the SIP Port and EXT SIP Port to 4000.

Daytona USA and Duke Nukem 3D may require a Jitter setting of Medium



ROUTER SETUP

10. You **MUST** login to your router, and do ONE of the following:

DMZ your VoIP adapter's IP Address

-or-

PORT FORWARD 5060 UDP and 16384 - 16482 UDP

IMPORTANT! - IF YOU CHANGED YOUR VOIP ADAPTER'S PORT NUMBER THEN YOU'LL NEED TO PORT FORWARD THAT NUMBER INSTEAD!

Info	System	SIP	Provisioning	Regional	Phone	Line 1	Line 2	User 1	User 2	User Login	Basic	Advanced
Line Enable: yes												
NAT Settings												
NAT Mapping Enable: yes												
NAT Keep Alive Enable: yes												
NAT Keep Alive Msg: \$NOTIFY												
NAT Keep Alive Dest: \$PROXY												
Network Settings												
SIP TOS/DiffServ Value: 0x68												
RTP TOS/DiffServ Value: 0xb8												
Network Jitter Level: low												
Jitter Buffer Adjustment: disable												
SIP Settings												
SIP Port: 5060												
EXT SIP Port: 5060												
SIP 100REL Enable: no												
Auth Resync-Reboot: yes												
Auth INVITE: no												
Auth MWI: no												
SIP Proxy-Require: no												
SIP Remote-Party-ID: no												
SIP GUID: no												
SIP Debug Option: none												
RTP Log Intvl: 0												
Restrict Source IP: no												
Referor Bye Delay: 4												
Refer Target Bye Delay: 0												
Referee Bye Delay: 0												
Refer-To Target Contact: no												
Sticky 183: no												
Call Feature Settings												
Blind Attn-Xfer Enable: no												
MOH Server:												
Xfer When Hangup Conf: yes												
Conference Bridge URL:												
Conference Bridge Ports: 3												
Proxy and Registration												
Proxy:												
Outbound Proxy:												
Use Outbound Proxy: no												
Use OB Proxy In Dialog: yes												
Register: no												
Make Call Without Reg: yes												
Register Expires: 3600												
Ans Call Without Reg: yes												
Use DNS SRV: no												
DNS SRV Auto Prefix: no												
Proxy Fallback Intvl: 3600												
Proxy Redundancy Method: Normal												
Voice Mail Server:												
Mailbox Subscribe Expires: 2147483647												
Subscriber Information												
Display Name:												
User ID: 11												
Password:												
Use Auth ID: no												
Auth ID:												
Mini Certificate:												
SRTP Private Key:												
Supplementary Service Subscription												
Call Waiting Serv: yes												
Block CID Serv: yes												
Block ANC Serv: yes												
Dist Ring Serv: yes												
Cfwd All Serv: yes												
Cfwd Busy Serv: yes												
Cfwd No Ans Serv: yes												
Cfwd Sel Serv: yes												
Cfwd Last Serv: yes												
Block Last Serv: yes												
Accept Last Serv: yes												
DND Serv: yes												
CID Serv: yes												
CWCID Serv: yes												
Call Return Serv: yes												
Call Redial Serv: yes												
Call Back Serv: yes												
Three Way Call Serv: yes												
Attn Transfer Serv: yes												
Unattn Transfer Serv: yes												
MWI Serv: yes												
VMWI Serv: yes												
Speed Dial Serv: yes												
Secure Call Serv: yes												
Referral Serv: yes												
Feature Dial Serv: yes												
Service Announcement Serv: no												
Audio Configuration												
Preferred Codec: G711u												
Silence Supp Enable: no												
Use Pref Codec Only: yes												
Silence Threshold: medium												
G729a Enable: yes												
G723 Enable: yes												
G726-16 Enable: yes												
G726-24 Enable: yes												
G726-32 Enable: yes												
G726-40 Enable: yes												
FAX Passthru Codec: G711u												
DTMF Process INFO: yes												
FAX Codec Symmetric: yes												
DTMF Process AVT: yes												
FAX Passthru Method: NSE												
DTMF Tx Method: Auto												
FAX Process NSE: yes												
Hook Flash Tx Method: None												
FAX Disable ECAN: yes												
Release Unused Codec: yes												
Dial Plan												
Dial Plan: (*xx)[3469]11[0]00[2-9]xxxxxx[1xxx[2-9]xxxxxx50]xxxxxxxxxxxxxx.)												
Enable IP Dialing: yes												
Emergency Number:												

SATURN US NETLINK SETUP

US NetLink games will ONLY connect with OTHER US NetLink games.
JP XBand games will ONLY connect with OTHER JP XBand games.
*(you cannot mix and match US / JP discs, as they will not communicate)

1. Put in any of the following **US NetLink** games into your Saturn:

- Daytona USA CCE NetLink Edition
- Duke Nukem 3D
- Saturn Bomberman
- Sega Rally Championship
- Virtual On: Cyber Troopers NetLink Edition

***If you and your opponent BOTH have US NetLink Modems (not JP Xband Modems) then you'll need to use the PATCHED US games.*

If at least ONE of you is using a JP Xband Modem, you can also use the RETAIL games. This is because the JP XBand modem sets the data rate to 14.4k, which provides a far superior performance over VoIP.

A US NetLink Modem can sync up perfectly with a JP Xband Modem that has set the data rate. When there is no JP XBand modem involved, use of PATCHED game images overrides and sets the 14.4k data rate between two US NetLinks modems.

PATCHED game images are available at <http://saturn.dreampipe.net>

2. When the game loads select "QuickLink".

3. Click on "Setup" on the main menu and you'll be brought to the following screen:

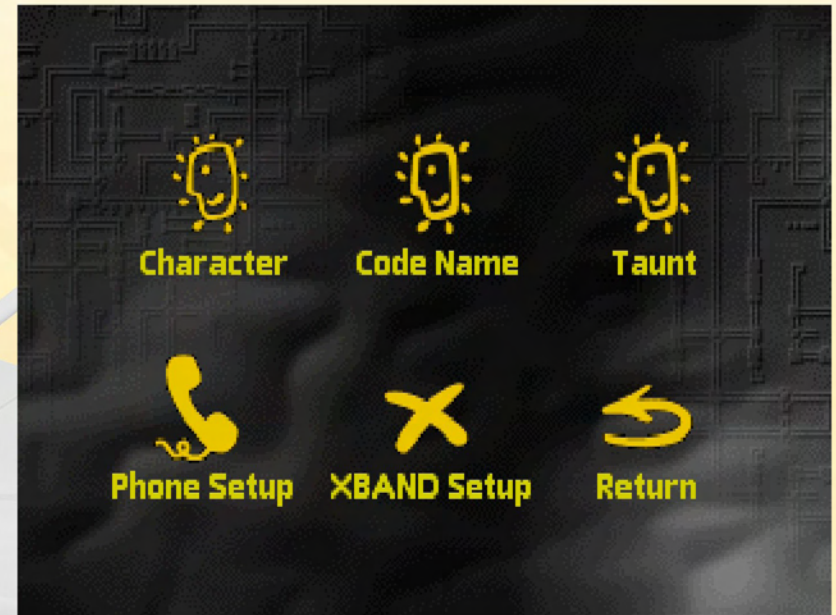
4. Click on "Character" select an Avatar.

5. Click on "Code Name" Type desired username.

6. (Optional) Click "Taunt" to put in some trash talk you want to show up before the match starts.

7. Click on "Phone Setup" select Tone.

8. Click Return.



HOSTING A NETLINK GAME

1. Simply click on "Wait", then you'll need to text message your opponent that you are ready for them to dial you. The option to practice or watch a slide show will appear while you wait.

DIALING INTO A NETLINK GAME

1. You will need your opponent's **Public IP Address**.
If they followed this guide then their UserID will be 11 and port will be 5060.
However these may vary depending on what user SIP Port was entered.
2. Before dialing your opponent, you should verify that they are "waiting" to be dailed.
3. To Dial, you will need to use the following format:

(UserID)*(IPAddress)*(PortNumber)#

NOTE:

For IP address, use an asterisk to separate the numbers where you would normally use periods. ####*####*####

EXAMPLE:

If you and your opponent have followed this guide, then both your UserID's will be **11** and ports will be **5060**. We'll assume that the public IP address for the person you are dialing is **127.1.1.0**

Under these circumstances, this is what you would dial: **11*127*1*1*0*5060#**



SATURN JP XBAND SETUP

1. Put in any of the following **PATCHED JP XBAND** games into your Saturn:

- Daytona USA Championship Circuit Edition for SegaNet
- Decathlete for SegaNet
- Puyo Puyo Sun for SegaNet
- Puzzle Bobble 3 for SegaNet
- Saturn Bomberman for SegaNet
- Sega Rally Championship Plus for SegaNet
- Sega Worldwide Soccer '98 for SegaNet
- Virtua Fighter Remix for SegaNet
- Virtual On: Cyber Troopers for SegaNet

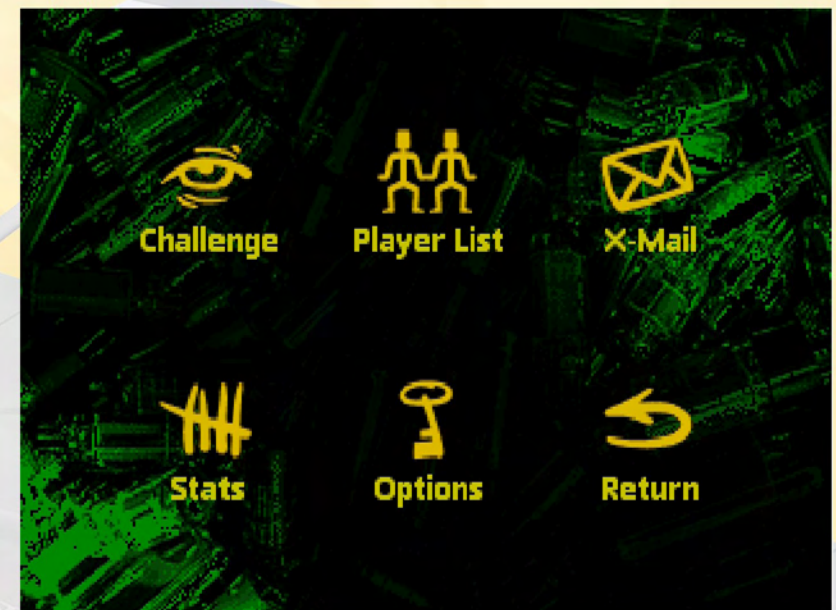
Patched games available at <http://saturn.dreampipe.net>

NOTE: Patched JP XBand games are **multiplayer only**. There is **NO** single player option.



HOSTING A JP XBAND GAME

1. Select an empty slot and name the Character "**s1**". (*this signifies you as the slave player being dialed*)
2. Select the challenge button, select yes to register and wait for a call.



DIALING INTO A JP XBAND GAME

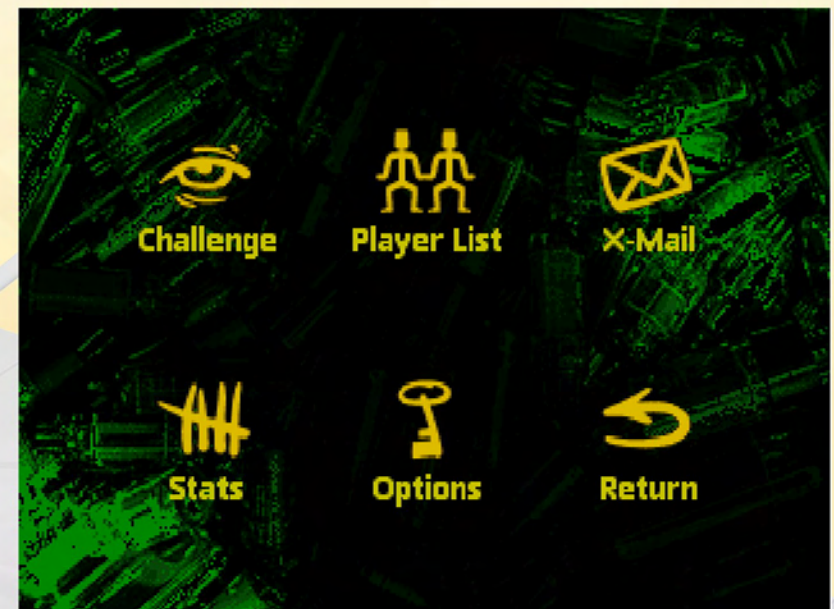
1. You will need your opponent's **Public IP Address**.
If they followed this guide then their UserID will be 11 and port will be 5060.
However these may vary depending on what user SIP Port was entered.
2. Login to your VoIP adapter, and make the following modification to the Dial Plan under the **Line 1** Tab at the bottom.

Dial Plan	
Dial Plan:	(*xx <9999:11@127.1.1.1:5060> [[3469]11 0 00 [2-9]xxxxxx 1xxx[2-9]xxxxxxS0 xxxxxxxxxxxxx.)
Enable IP Dialing:	yes <input type="button" value="Emergency Number:"/>
<input type="button" value="Undo All Changes"/> <input type="button" value="Submit All Changes"/>	

Replace the **127.0.0.1:5060** with your opponent's IP address & Port number

(*xx|<9999:11@127.0.0.1:5060>|[[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxxx.)

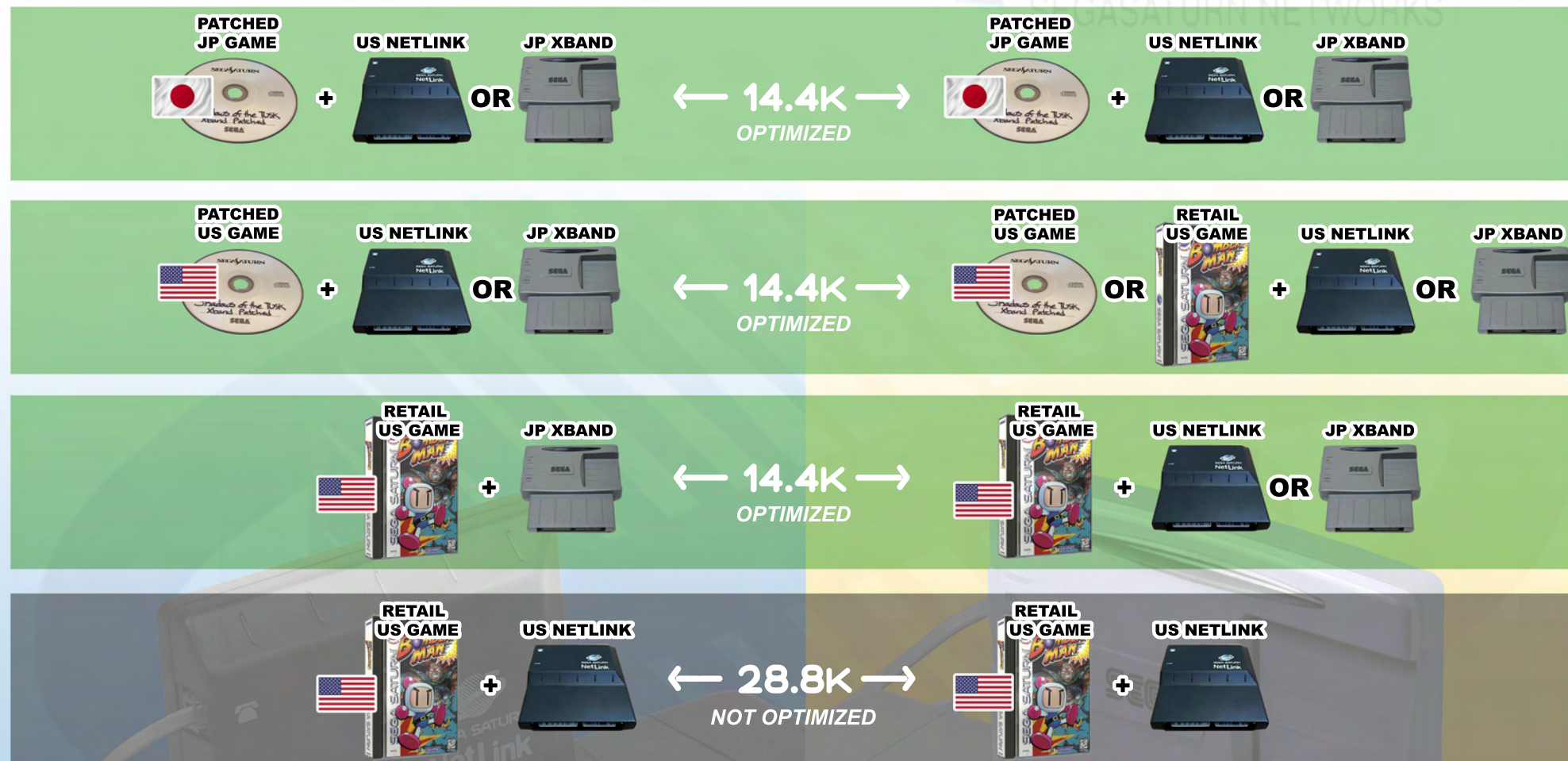
3. Select a empty slot and name the Character "**m99999#**"
4. Before dialing your opponent, you should verify that they are "waiting" to be dialed.
5. Select **Challenge** when your opponent is ready, and select **yes** when asked to **register**.



XBAND WAS DESIGNED FOR 14.4K...

Believe it or not, Catapult actually designed their XBand modems to work almost flawlessly at 14.4k modem speeds. They fine tuned each and every game for this data rate. The US NetLink's 28.8k speed was really only useful for browsing of the internet, as it was marketed in the US as an internet appliance.

14.4k modem speeds have been found to work **better over VoIP** without any negative impacts to game play. To connect at this speed, a JP XBand modem must be used on at least one side of the connection when playing US RETAIL games. When using **PATCHED** game images, all modems and games will connect at 14.4k regardless of configuration. Here is a visual chart showing all possible match-up configurations:



ACCESSING THE NETLINK ZONE

NetLink Zone is only available on USA NetLink Games.

1. Insert any Sega Saturn PlanetWeb Disc
2. Setup your saturn following the DreamPi setup guidelines for Dreamcast.
Instructional Video: <https://www.youtube.com/watch?v=VgRLnj2YR3s>
3. Boot up any NetLink game and click on the NetLink Zone When you connect your Saturn via Dreampi your game will automatically redirect to the appropriate site no modifications necessary.

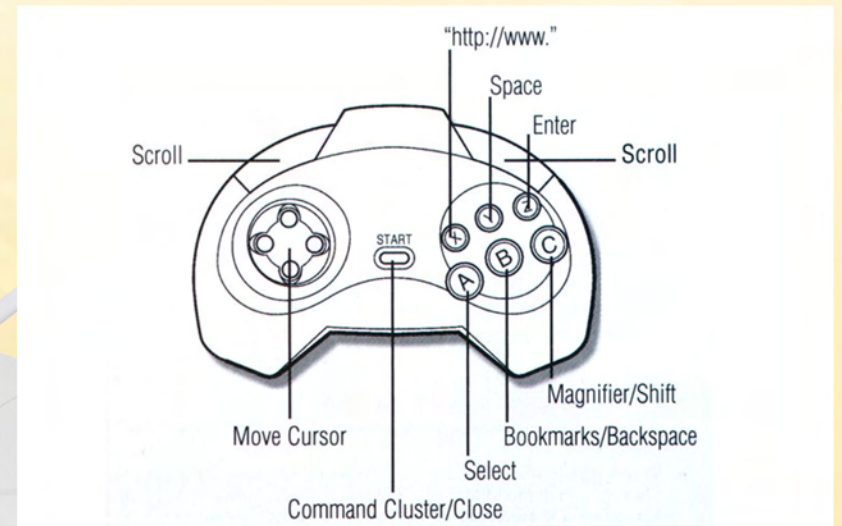
****Alternatively you can set your primary DNS to: 46.101.91.123 if your dial up IP allows this.***

4. Features:
 - In browser Quick Link direct launch
 - IRC Chatroom
 - Guestbook with Registration
 - Email
 - Help Site
 - Direct Link to Saturn.Dreampipe.Net



SEGA SATURN NETLINK UTILITIES

1. SaturnPipe Homepage
<http://saturn.dreampipe.net>
2. Planet Web Browser 4.0
<http://www.mediafire.com/file/61ics5rjs2ve2ih/planetweb4035b.rar>



ACCESSING STCC LEADERBOARDS

SEGA TOURING CAR CHAMPIONSHIP LEADERBOARDS COMING SOON!

TROUBLESHOOTING TIPS

GAMES FREEZE UP CONSTANTLY...

- Set Jitter to Medium
- Double check that ALL settings for you and your opponent MATCH EXACTLY.
- You should only play with opponents from the same continent. Overseas matches unfortunately are not currently supported.
- Only use wired connections. Wireless connections will usually always fail to connect.

WHEN SOMEONE DIALS ME, NOTHING HAPPENS ON MY SCREEN...

- Verify your VoIP is DMZ'd or is port forwarded properly.
- Try setting your sip and sip ext port to 4000
- Check your stun server settings make sure everything is typed in correctly
- Check your Gateway and make sure it's correct
- Make sure your public IP address hasn't changed on you since your last connection attempt.
- When you are in Wait Mode the green light on the VOIP adapter will NOT be on.
- When you are in Dial Mode the green light on the VOIP adapter WILL be on.

HOW DO I FACTORY RESET MY VOIP ADAPTER?

- To perform a factory reset on your SPA-1001 dial **** on the phone. You should hear a Sipura message asking you to enter your selection. Then dial 73738#

I CAN'T GET PAST THE "ANSWER" SCREEN ON A NETLINK CALL...

- Double check your VoIP adapter is **DMZ'd** or has **Port Forwarding** enabled on the **correct ports**.
- On **SIP Tab**, disable enable Stun Server. On **Ext IP** put your **ISP ip address** in there and save.
- Try switching around who dials, and see if that fixes the problem.
- Try changing your port on the VOIP adapters **from 5060 to 4000 (Both users should do this)**
- Try switching **STUN Server to stun.wtfismyip.com**
- Test if your Internet Connection can handle VOIP: http://www.whichvoip.com/voip/speed_test/ppspeed.html

WHEN I DIAL MY OPPONENT, IT IMMEDIATELY SAYS THEIR LINE IS BUSY...

- Both you and your opponent need to unplug your voip adapters for about 10 seconds and then plug them back in, and try again, this almost always fixes the issue.

HOW DO I ACCESS THAT COOL TAUNT SCREEN ON JP XBAND GAMES?

Type mp9999# or sp to show taunt screens for JP games. Taunt messages do not save after restart.

