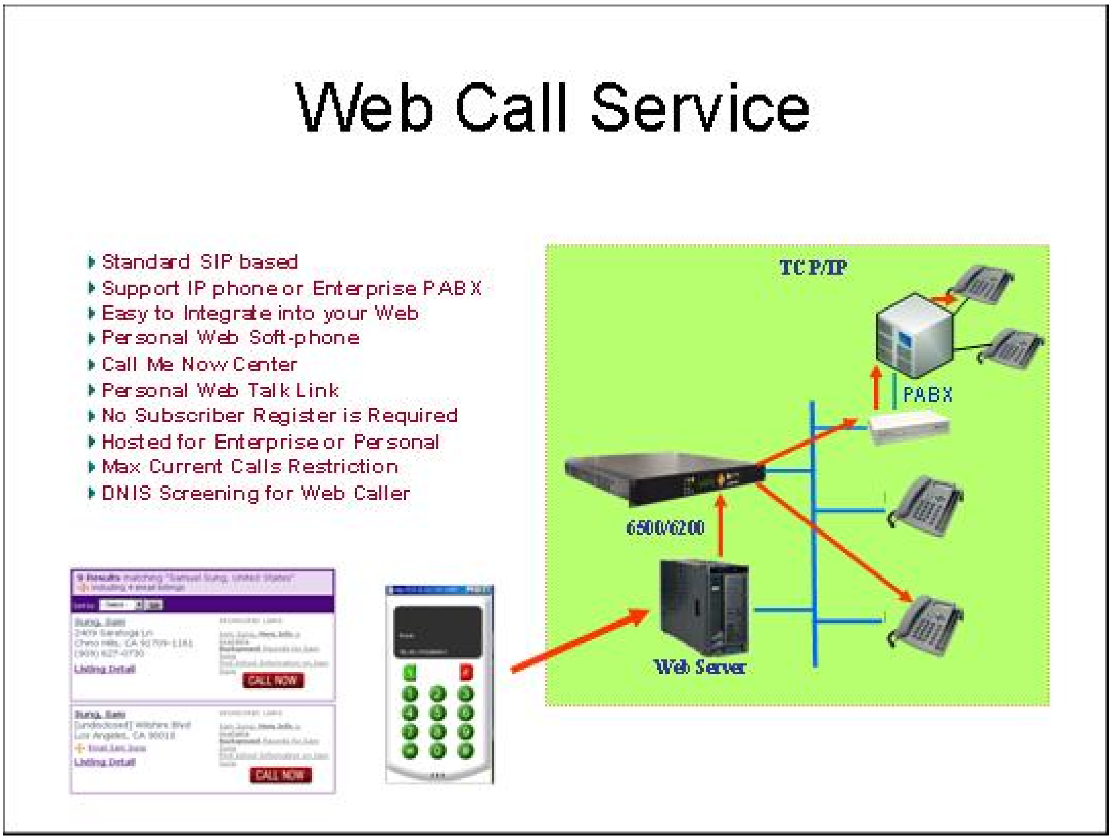
The Implementation Guide for Welltech Web Call Service

Target Objective:

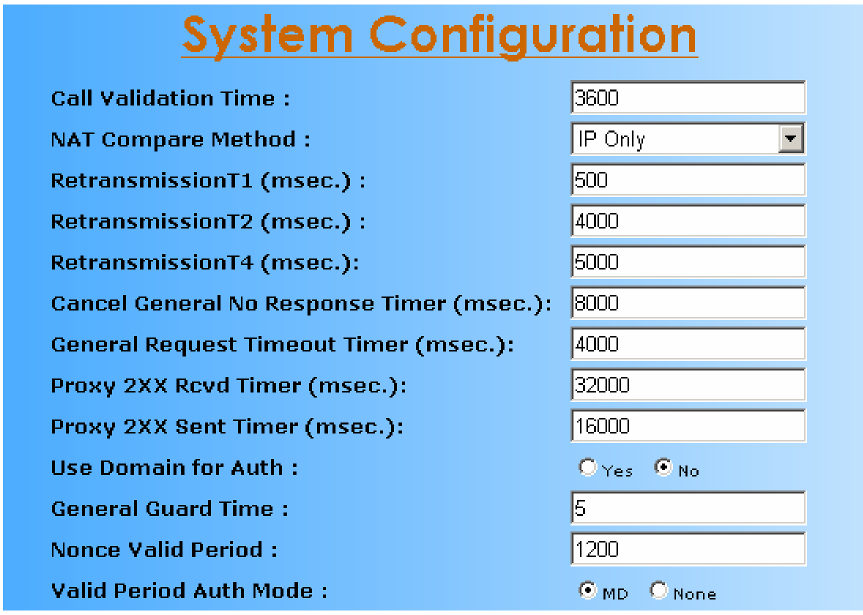
Welltech Web Call solution provides an easy way to be integrated into your Web home page to provide a free call from your customer or supplier to your business. It doesn't like 800 caller free-calls which the enterprise pays for it. The Welltech Web call solution using the VOIP to provide you a truly free 800 call services. It can be used for call center, service center, help desk or sales leads.



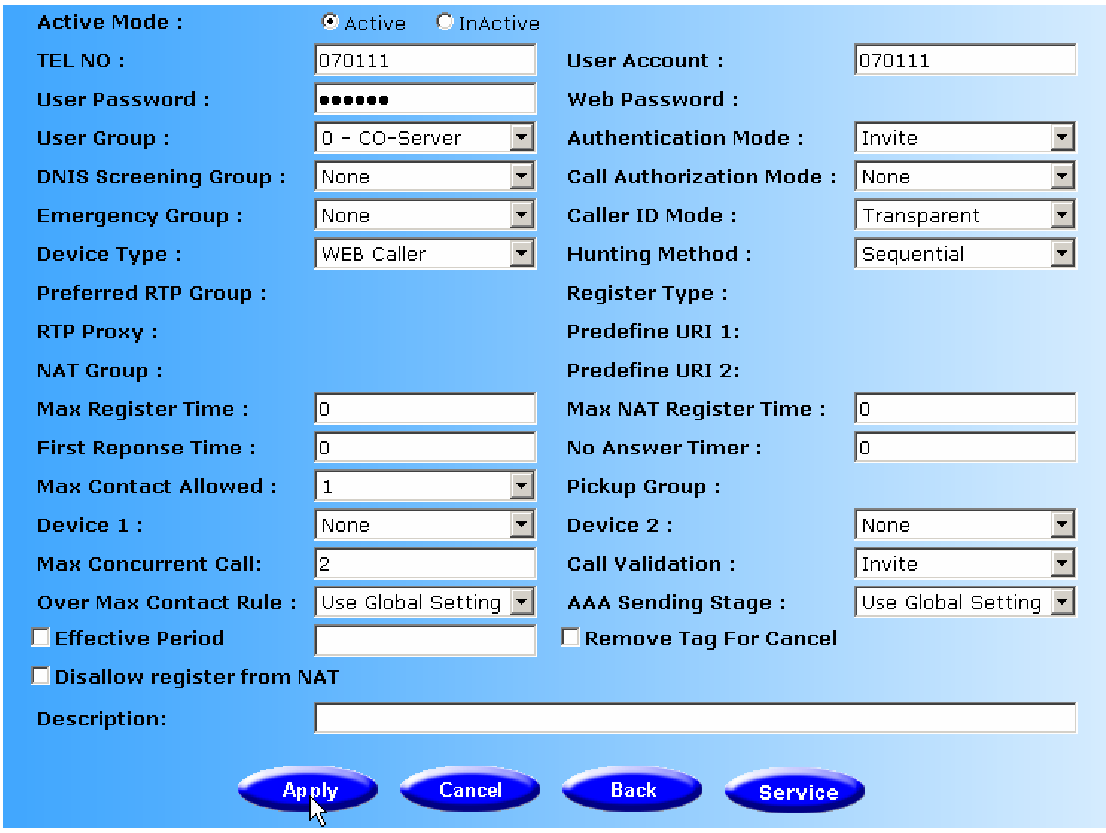
Recommendations:

WellSIP 6500/6200 Settings:

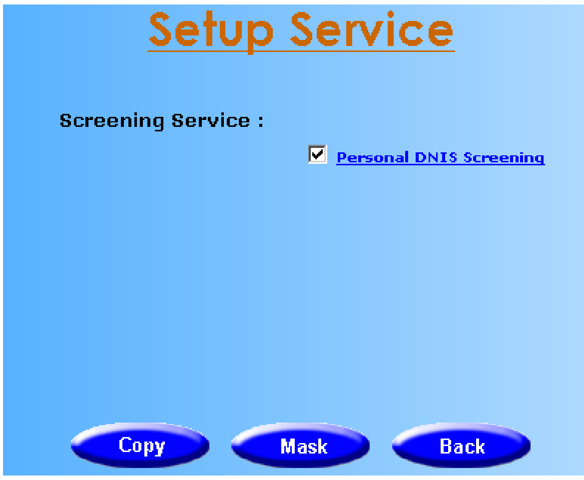
1. Make sure you have got Web Call license from Welltech
2. System Parameter Changes:
   - System -> Advance -> Valid Period Auth Mode: MD



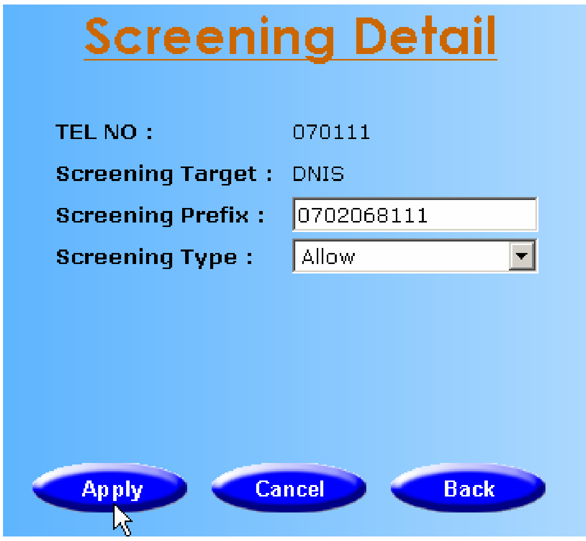
3. Create a Web Caller Subscriber (e.g. 070111)
   - Subscriber TEL: 070111
   - Subscriber User ID: 070111
   - Subscriber Password: ***** (please make sure you give a random password to have strong enough security)
   - Subscriber Type: WEB Caller
   - Authentication Mode: Invite
   - Max Concurrent Call: 2 (2 simultaneously calls for this customer)
   - Apply



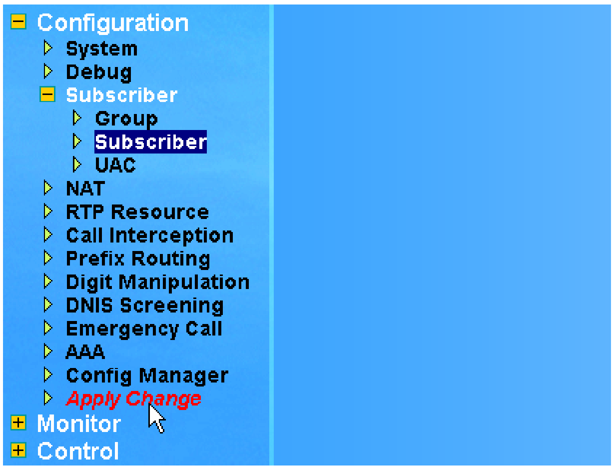
- Click Service



- Enable Person DNIS Screening and add the required DNIS Screening (e.g. 0702068111 allow). It will allow call 0702068111xx only. It is recommended to add the screening for each web call customer.
- Apply



4. Apply Change



Sample Code Files included:

Demo.jsp: main demo page to show the different company to call
Dialing.jsp: load Web call component and start the call
Sipocx.cab: Web Call OCX to be downloaded to IE
Status1.js: Java Script include file for call event handling
pic: the directory contains the sample pictures

Minimum changes are required to run the demo jsp:

Parameters need to be changed for web caller settings:

- String TEL No = "00002"; //web caller user id
- String Password = "00002"; //web caller password
- String Server IP = "192.168.19.77"; //proxy server IP
- String Server Port = "8085"; //proxy server UDP port (it is recommended to use 8000-9999)

Calling dial function to run dialing.jsp. The parameter is called party number:

- Dial(Tel)

The API Reference for Web Call Component 1.0.x

1. Property

1.1 Apply To [mandatory]

Set OCX to become the web caller mode
Long Apply To
1: Web Caller

1.2 Codec Priority [mandatory]

Set the priority of the codec of RTP payload
BSTR Codec Priority

g.729: Supported & recommended used for web caller
g.711Alaw64k: Supported (not recommended)
g.711Ulaw64k: Supported (not recommended)
g.723.1: Optional (contact Welltech if need)

For multiple codec selection, sing comma as the delimiters (e.g. g.729, g.711Alaw64K)

1.3 EchoLen

Set the tail length of the AEC (128ms-512ms)
Short EchoLen
Unit: ms
Default: 256ms

1.4 MicVol Level

Adjust the audio-level of the microphone
Short MicVol Level
Unit: dB
Range: -20 db ~ +20 db, default is 0 (0db)

1.5 Network Interface Index

Set the sipocx using which network interface on the computer
Short Network Interface Index
Default: 0 (for most of case)

1.6 Password [mandatory]

The web caller's password on WS6500/6200
BSTR Password

1.7 PayLoad Len [mandatory]

Set the length of the RTP payload
Long PayLoadLen
Unit: Bytes
G729/G.711: 20/40/60 (20ms is recommended)

1.8 Proxy IP [mandatory]

The 6500/6200 server's IP
BSTR Proxy IP

1.9 Proxy Port [mandatory]

The 6500/6200 server's SIP Service Port
Short Proxy Port

1.10 Secured

Enable Welltech Encrypted/Decryption for SIP and RTP.
boolean Secured
Default: false

If you turn on secured option, please register to WS6500/6200 encryption port (NOT regular port)

1.11 SpkVol Level

Adjust the audio-level of the speaker
Short SpkVol Level

Unit: dB
Range: -20db ~ 20db, default is 0 db

1.12 Tel Event Payload Type [mandatory]

Set the payload type for DTMF relay (RFC 2833)
Short Tel Event Payload Type

101 is recommended

Only apply when User Input Type is 0 (RFC 2833)

1.13 Tel No [mandatory]

WS6500 register telephone number
BSTR Tel No

1.14 User Input Type [mandatory]

Set the transmit method of the DTMF

Short User Input Type

0: RFC2833 (default and recommended)

2: SIP INFO

1.15 User Name [mandatory]

WS 6500 Web caller User Name

BSTR User Name

Method

2.1 BusyTone

Set the busy tone file
boolean Busy Tone(BSTR lpWaveFile)

Parameter:
lpWaveFile: The Busy Tone play back file name. The file must be a Windows Wav format based on Liner PCM 11KHz Mono.
Return:
True: success
False: failure

2.2 Cancel Call

Cancel an outgoing call (used only in ringing state)
boolean CancelCall(short sChannelID)

Parameter:
sChannelID: The channel ID returned by Make Call method.
Return:
True: success
False: failure

2.3 Channel Status

Get the channel status
long Channel Status(short sChannelID)

Parameter:
sChannelID: The channel ID returned by Make Call method.
Return:
Channel Status (refer to Appendix 1)
STATE_ERROR: Channel ID doesn't existed or failed to get the channel status

2.4 ndWeb

Terminate and unload the sipocx
boolean EndWeb()

Parameter:
None

Return:
True: success
False: failure

2.5 GetUSBVolume

Get current device voice volume
short GetUSBVolume(long iDevice)

Parameter:
iDevice:
0: speaker
1: microphone.
Return:
The voice volume of the selected device

2.6 HangUp

Hang-up a connected/Failed call
boolean HangUp(short sChannelID)

Parameter:
sChannelID: The channel ID returned by MakeCall method.
Return:
True: success
False: failure

2.7 HoldTone

Set the music file for hold state which is wav file based on Liner PCM 11KHz Mono.
boolean HoldTone(BSTR lpWaveFile)

Parameter:
lpWaveFile: The hold tone WAV file (full file name with path)
Return:
True: success
False: failure

2.8 Init

Initialize the sipocx
long Init()

Parameter:
None
Return:
-1: failure
0: success

2.9 KeepSIPSession

Send an OPTION message periodically. It is recommended to set the interval to 30-50 seconds in order to get through NAT traversal.
boolean KeepSIPSession(short sInterval)

Parameter:
sInterval: 0: disable
It is recommended to set to 30-50 seconds.
Return:
True: success
False: failure

2.10 MakeCall

Make an outbound call
Short MakeCall(BSTR sDNIS)

Parameter:
sDNIS: Called Party Number
Return:
>=0: The channel identifier
-1: failure

2.11 PlayKeyWave

Play a DTMF tone to caller
boolean PlayKeyWave(short sKeyNum)

Parameter:
sKeyNum: 0-9: DTMF Digits
10: *
11: #
Return:
True: success
False: failure

2.12 RingBackTone

Set the ring back tone file when remote party is ringing.
boolean RingBackTone(BSTR lpWaveFile)

Parameter:
    lpWaveFile: The ring back tone WAV file (full file name with path)

Return:
    True: success
    False: failure

2.13 SendUserInput

Send a DTMF out to remote party (RFC 2833 or SIP INFO)
Long SendUserInput(short sSendUserInput, short sChannelID)

Parameter:
    sSendUserInput:
        0-9: DTMF Digits
        10: *
        11: #

sChannelID: The channel ID returned by MakeCall method.
Return:
    0: success

2.14 SetUSBVolume

Set the device volume
void SetUSBVolume(long iDevice, short nNewValue)

Parameter:
    iDevice:
        0: speaker
        1: microphone
    nNewValue: voice volume
Return:
    None

2. Event

3.1 CallBusy

The destination user is busy
void CallBusy()

3.2 CallConnected

The Call is connected
void CallConnected()

3.3 CallDroped

The call is disconnected
void CallDroped()

3.4 CallForbidden

The outbound call is forbidden by proxy.
void CallForbidden()

3.5 CallHold

The call has been put on hold state
void CallHold()

3.6 CallNoAns

No answer from remote party
void CallNoAns()

3.7 CallNotFound

Wrong destination number
void CallNotFound()

3.8 CallRingBack

The remote party is ringing.
void CallRingBack()

3.9 CalledNotOnline

The remote party is temporary unavailable
void CalledNotOnline()

3.10 MicVolumeChanged

The microphone voice volume has been changed
void MicVolumeChanged()

3.11 SpkVolumeChanged

The speaker voice volume has been changed
void SpkVolumeChanged()

3.12 NetWorkError

Network error
void NetWorkError()

3.13 Ready

The web call is ready to use
void Ready()

4 Appendix 1

4.1 STATE_ENUM

```
STATE_HANGUP = 0,//0
STATE_HOLD,//1
STATE_CANCEL_HOLD,//2
STATE_HOLDING,//3
STATE_CALL_BUSY,//4
STATE_READY,//5
STATE_CANCEL_CALL,//6
STATE_CALL_MAKING,//7
STATE_CONNECTED,//8
STATE_DISCONNECTED,//9
STATE_CALL_INCOMING,//10
STATE_CALLED_NOT_ONLINE,//11
STATE_CALL_FORBIDDEN,//12
STATE_CALL_NO_BALANCE,//13
STATE_CALL_NOT_FOUND,//14
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STATE_ERROR,//15
STATE_TRANSFERHOLDING,//16
  STATE_TRANSFERHOLD,//17
STATE_REFER_ACCEPTED,//18
STATE_GET_VOICE_MAIL,//19
STATE_FUNCTION_KEY_SUCCESS,//20
STATE_GET_CONF_MEMBERS,//21
STATE_CONF_OUT_OF_RESOURCE,//22
STATE_CONF_MEMBER_BUSY,//23
STATE_CONF_MEMBER_NOT_FOUND,//24
STATE_CONF_MEMBER_FORBIDDEN,//25
STATE_UNREGISTER_OK,//26
STATE_CONF_MEMBER_NO_ANSWER,//27
STATE_BAD_REQUEST,//28
STATE_CALL_NOANS,//29
```