XPression
TDB / Acoustic Modelling questions

> Is the XPression a sampler? Or is it more of an advanced "ROM-pler" because the user cannot sample?

The XPression is not a sampler! It does not record samples! But it's not really a "ROM-pler" either. Even though it uses samples the sound generation / manipulation is different and much more advanced compared to typical "ROM-plers" or samplers.

On top of it since software version 1.2 there is also an analogue synth onboard which can be layered with the samples.

> From what I understand TDB is what gives the XPression more expression than other samplesynths. I like the unpredictable acoustic parameters that are in Yamaha's VL physical modelling synthesis. I've been trying various ways to reproduce it in a sampler, but it's not the same. I'm interested in learning more about TDB...

The idea of TDB / Acoustic Modelling is rather simple:

The goal is first to analyse the deviations from a theoretically "perfect" sample in different dynamics and with different articulations. Then recreate these changes with digital / analogue means and apply it to this (formerly recorded) *"perfect"* sample.

Instead of using samples that have certain "sound characters" (like clean/dirty) or articulations (staccato/legato) TDB needs perfectly clean, uncoloured samples. These samples incorporate very fast and absolutely even attacks. There shouldn't be any extra overtones audible within the first 20 – 50 ms. The overtone spectrum has to be at a perfect "forte" level producing as many overtones as possible without "distortion" (like in a brass fff or a buzzy overblown sax). All samples need to be perfectly equal since all manipulation which are done later will be multiplying any difference in the original sound. These samples are extremely hard to create. It usually takes 2 hours per sample. (Low C / C# on the flute took two full days each.)

Now, after obtaining these basic uncolored *"perfect"* samples all variations of sound, uncleanness of sustained sound or attacks are being modulated into the sound by synthesis. Otherwise the instrument would sound completely boring and dead (as it is the case with most traditional samplers)!

There is a multi-band fully parametric EQ that is set up to follow dynamics, pitchbend and vibrato individually for each band. This recreates (partially) the different sound spectrums of the different dynamic and pitch levels (this process could be compared to "formant synthesis" which is used in Physical Modelling). There are several distortion modules with variable distortion amount (and filters to tune the overtones) in order to create harmonic overtones which are not present in the original sample. Of course there is your standard low pass filter which follows pitch & volume with it's frequency, mix from the dry to the filtered sound and Q factor. There is also a keyfollow high pass filter that can filter some unwanted low noise originating in the very close micing during sampling. There are two notch filters that you can tune right into any of the first 16 overtones to make it e.g. more clarinet like by reducing the 2nd and 4th overtone, make the sample more "fat" by boosting the fundamental – or anything else you like.

Now, one of the most interesting parts is that you can create all kind of additional modulations by adding oscillators with frequencies other than your "clean" sound source to produce partials that don't necessarily have to be harmonic. This is important as in real live even the best players play one out of a hundred notes absolutely "perfectly" in the sense of perfect attack with no other overtones, etc.
Such oscillators can be setup to follow the overtone series as if they were produced by the different original instruments. That means not just a fixed interval. The pitch, envelope and volume are controlled by the pitch that is being played as well as the volume/velocity and random. And it can be limited to a minimum/maximum range as well. When attacking a note there might be an unpredictable overtone above or below the pitch that you are actually playing. There is a number of these modulations. The attack is treated separately from the sustained part. Growl obviously needs different frequencies than multiphonics. (BTW multiphonics can be set to auto starting at a certain pitch up.) There is another modulation which is called "harmonic dirt". These produce mostly the additional lower octave when playing in the second octave of e.g. a sax (this creates a typical old "Sanborn" sax sound).

Most of these transformations behave differently on legato/non legato. Several additional envelope generators for attacks (on top of the standard envelopes) can be used for extra punch or smoothness.

These extra oscillators are then modulated into the sample by using e.g. amplitude modulation which sounds basically as if you would combine different frequencies inside the tube of a wind instrument. On top of it there is noise (like e.g. breath samples) that can be added or rather modulated into the sound also (again differently for the attack or sustain).

All of these parts have their own behaviour on dynamics/pitch and have certain random ranges. Some of them also modulate each other which results in a slightly more complex algorithm.

That's the reason why the XPression is capable of repeating the same sample many times while still not sounding repetitive (there is no "machine gun" effect). This gives it also the flexibility to change the sound drastically without the need of multiple samples for different sound types, articulations or volume levels.

This sample modulation can get quite complex – although all components by themselves are rather simple. The hard thing is to balance all of these parameters to get a realistic sound and a good feeling playability.

What makes XPression sound better than other "samplers" is the use of many random parameters. With most other sampled instruments you might not be able to e.g. double two tenor saxes. You might run into real bad phasing problems. Here you can add a variation by offsetting the samples by +/- 1 semitone and add random pitch (separately for attack, fast and slow sustain) and a random delay while setting up the pitchbend range individually for each of the horns. Defining different dynamic curves/ranges will have them even behave differently to the level that you are playing.
This means that even if you play a five part horn section at once (with the XPression's chord mapper) all instruments will keep their own identity.

> Why is it so important that the samples are recorded so precisely?

One of several reasons why all sample attacks have to have the exact same timing originates partially in the legato mode and other note transitions like falls. When playing legato the first couple of milliseconds of the sample are cut off and replace by an attack envelope while there is a short release of the last sample so there is no gap in between notes. There has to be just a tiny little bit left of the attack in order for it to sound real. If it's too much it doesn't sound legato if it is not enough it sounds like a synth. There is a also very fast pitch envelope during the transitions which won't sound correctly when the attacks are not optimized. Each instrument needs its own legato transition parameters. By "slightly" varying them you can change the legato sound quite a bit. E.g. a clean classical sax or a roughly articulated jazz sax.

If you would use samples that don't have the exact same attack time the legato would sound different between different intervals / notes. Variations in the behaviour could not be set precisely or would behave unpredictably. Generally articulations would not be controllable as expected.

Another reason is that imperfect attacks are recognizable and can produce the so called "machine gun" effect when repeating the same note several times.

> What's the difference between Physical Modelling and XPression's Acoustic Modelling or TDB?

Physical Modelling synthesizes the sound of an acoustic instrument by splitting it (simplified) into an "exciter" (e.g. in case of a saxophone: the reed) and a "resonator" which represents the body of the instrument. There is a feedback loop between the "resonator" and the "exciter" which can change the behaviour of the "exciter" according to the resulting sound in the resonator. There are many different sub categories of equations used to actually calculate the sound. Strings use different algorithms than winds or drums.

Physical Modelling makes it possible to highly influence the sound by changing a large number of parameters controlling how the "exciter" is behaving. Usually the "resonator" has fixed parameter settings which are representing the size, form and material of the body. This means you can change the way how the bow is getting the string on a violin to sound by modifying the bow pressure or speed or using different damping factors for the material of the string but you cannot change the body of your violin.

The problem here is obvious: In order to achieve such flexibility the calculations have to be highly simplified. A cheap student violin might be a copy of a Stradivarius and have the exact same dimensions. There is no way the equations used by Physical Modelling could be precise enough to tell the sound difference between the two. It is definitely not possible to make a distinction between different shapes of instruments or e.g. a curved or straight soprano sax.

Physical Modelling's sound production is very flexible and will sound rather similar to a real instrument but never as complex and rich as a real instrument. You will recognize the sound is meant to be a saxophone and playing it might feel as musical as playing a real saxophone. But you would not be able to model an exact mouthpiece/neck. And how would you be able to create a difference between two brands of saxophones if the models of the resonators are only one or two-dimensional instead of three-dimensional?

The XPression’s Acoustic Modelling uses samples as an “exciter”. This means there is no way to influence the “exciter” as the actual samples themselves cannot be changed.

The idea behind the XPression’s sample manipulation is that there is an optimal sound that can be produced with a certain instrument setup. In case of a saxophone this would be a certain reed, mouthpiece and instrument combination. The optimal sound is the loudest and most open sound creating the maximum amount of overtones without overblowing (distorting) the instrument. It creates a perfect representation of the entire instrument – and only of this particular instrument! More high quality instruments of the same type (e.g. different brands of saxes or mouthpieces) can be added easily by adding new sample sets later.

As a result the samples represent both “exciter” and “resonator” in terms of a Physical Modeling instrument.

At first this does not sound very flexible at all – and in fact many samplers or sample players stop here and use rigid keyswitches to play different dynamics or articulations. This makes it pretty much impossible to do smooth transitions between various dynamic levels or sound variations – such as growl, multiphonics or subtone. Not even to mention the limited number of articulations, dynamic levels and inconveniences like predefined timing of crescendos, and so on . . .

So how does the XPression do it?

The XPression uses an optimal captured sound representation of the instrument and calculates the difference in the overtone spectrum and attack behaviour for the various dynamic levels or sound variations in realtime. It can subtract frequencies while modulating (e.g. breath) noise into the sound (which actually will not sound like air but again mainly changes the overtone spectrum). It can add non existing frequencies to create additional harmonic or non harmonic overtones or noise components in loud or rough passages. It can change the attack behaviour of different components according to the dynamic or legato status. Midi CC messages can drastically alter the sound by adding non harmonic frequency components. Since real instruments cannot mechanically reproduce the exact same sound over and over XPression also adds random changes to all of these (and many more) parameters.

Other than with traditional samplers all of these elements are not simply layered on top of the sample but they are calculated <u>into</u> the sound. This is why the XPression has an unbeatable dynamic behaviour and gives you a musical flexibility that could only be achieved with Physical Modelling instruments up until now.

Conclusion: the difference between the XPression’s Acoustic Modelling and Physical Modelling instruments is that the XPression simply sounds better while having an equally excellent playability and response!